



SIP Firmware Release 2.0 for IP Phone 1140E Administration

Document status: Standard
Document version: 03.09
Document date: 15 September 2008

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New in this release

The following changes are introduced for *SIP Firmware Release 2.0 for IP Phone 1140E Administration* (NN43113-300).

Server Supplied User Credentials

This release adds support for downloading a users login credentials from a central repository. The login credentials are associated with the MAC address of the set. This feature simplifies installation of the IP Phone by allowing it to automatically log into the SIP proxy server without end user intervention.

To provision this feature see "[Set configuration commands](#)" (page 60).

Expansion module for the IP Phone 1100 Series when connected to IP Phone 1100 series SIP Client

An expansion module providing 18 additional feature keys, can be attached to a Nortel IP Phone 1140E with SIP firmware. Up to three expansion modules can be added to the phone. The feature keys in the expansion module provide the same functionality as the feature keys on the IP Phone.

Additionally, this feature adds the ability to automatically populate the extra feature keys, using the friends list or the address book as sources.

For more information on the Expansion module refer to, *SIP Firmware for IP Phone 1100 Series Expansion Module User Guide* (NN43110-301).

To provision this feature see "[Feature configuration commands](#) " (page 42).

IEEE 802.1ab Link Layer Discovery support

Discovery protocols provide a mechanism to identify devices attached to a network. Popular network-management systems use automated discovery to obtain the topology of a network. These applications detect adds and removals, provide Layer 3 information, and group the attached devices into IP subnets.

IEEE has developed 802.1ab Link Layer Discovery Protocol (LLDP), a standard for discovering the physical topology between neighboring devices. 802.1ab LLDP defines a standard method for Ethernet network devices such as switches, routers and IP Phones to advertise information about themselves to other nodes on the network and store the information they discover in a MIB.

This feature implements 802.1ab and its IEEE 802.1/802.3 and TIA-MED organizationally specific extensions.

To provision this feature see ["802.1ab Link Layer Discovery Protocol" \(page 99\)](#) and [Procedure 13 "Provisioning the Device Settings parameters" \(page 111\)](#). The parameters in [Table 18 "TLV formats" \(page 100\)](#) can be configured on the IP Phone.

IP Phone 1140E with SIP firmware PRACK and UPDATE support feature

The combined support of PRACK and UPDATE allows the IP Phone 1140E with SIP firmware to provide reliability to provisional responses, and the ability to update session parameters during call setup as well as after the initial invite has received a final response. The combination of reliable provisional responses (PRACK) and the ability to change session parameters before call establishment (using UPDATE) will improve the IP Phone 1140E interactions with some PSTN networks where the parameters of a session may need updating before the call is established.

PRACK is used for adding reliability to provisional responses (such as 180 Ringing) in SIP messaging. The PRACK request plays the same role as ACK, but for provisional responses. [RFC 3262]

UPDATE allows a client to update parameters of a session (such as the set of media streams and their codecs) but has no impact on the state of a dialog. In that sense, it is like a re-INVITE, but unlike re-INVITE, it can be sent before the initial INVITE has been completed. This makes it very useful for updating session parameters within early dialogs. [RFC 3311]

To provision this feature see ["Feature configuration commands " \(page 42\)](#).

RFC 3581

This feature implements an extension to SIP for Symmetric Response Routing for the IP Phone 1140E with SIP firmware. This extension permits the conduction of SIP dialogs through a Symmetric Network Address Translator (NAT) using UDP.

This allows the phone to work from behind and/or in front of a symmetrical NAT with servers and/or clients that support RFC 3581. In particular, it enhances the capability of the IP Phone 1140E with SIP firmware to interoperate with other proxies and to work in any network configuration. For this feature to work properly, the receiving end device must support RFC 3581.

To provision this feature see ["Feature configuration commands " \(page 42\)](#).

RTP port configuration

This RTP port configuration feature provides options to set or change RTP ports in the Nortel IP Phone 1140E with SIP firmware. RTP port configuration is available only through the provisioning server.

To provision this feature see ["Feature configuration commands " \(page 42\)](#).

Voice Quality Monitoring

When Voice Quality Monitoring (VQMon) is enabled, the IP Phone 1140E with SIP firmware gathers statistics regarding the quality metrics of the current call and sends reports to the server at regular intervals. The call-related statistics contain condensed information about the SIP Session Description Protocol (SDP), the Call ID, the local and remote address, voice quality-related statistics, Zulu times for start-time and the time the report was sent. The voice quality-related statistics include jitter, packet loss, delay, burst gap loss, listening R-factor, R-LQ, R-CQ, MOS-LQ and MOS-CQ. The reports can be used for QoS monitoring. A third party server is required to collect and organize the data.

End of call reports are sent with the Zulu time indicating start of session and end of session; session interval reports are sent with interval defined in the configuration using time stamps, and alert reports are sent with current time stamps with a limited frequency of 1 report per 5 seconds if any alert violation persists.

Release 2.0 supports interworking with SQLMediator 1.1.

To provision this feature see ["VQMon configuration commands" \(page 57\)](#).

Distinctive ringing and Call waiting

The Distinctive ringing feature permits users to distinguish between different types of call actions by playing a different ringing pattern. The request for a specific ringing pattern comes from the server at the time the call is being established. The Nortel IP Phone 1140E with SIP firmware does not request the playing of distinctive ringing from other parties. The ringing patterns to be used follow the North American standards which also include call waiting

tones for the times when the receiving end is already engaged in a call session. The predefined ringing pattern identifiers available on the phone accommodate usage by the BroadWorks servers.

To provision this feature see ["Feature configuration commands " \(page 42\)](#).

Shared Call Appearance

The Shared Call Appearance feature allows a given line to be configured with multiple locations, essentially allowing multiple endpoints to login to the system while using the same external number. Any one of these locations can be used to originate or receive calls. Any call appears to the other party to be originating from or terminating to the same number, regardless of the location initiating or receiving the call.

The Shared Call Appearance feature allows a user to pick up a call that was put on hold by another user of the same group and it allows a user to join an active call of another user in the group.

The Shared Call Appearance feature is supported on BroadWorks servers.

To provision this feature see ["Feature configuration commands " \(page 42\)](#).

Transfer to Voice Mail

The transfer to Voice Mail feature enables a soft key to transfer an incoming call, or a call waiting, to a voice mail box on the IP Phone 1140E with SIP firmware.

The transfer to Voice Mail feature is supported on MCS 5100 servers only.

To provision this feature see ["Feature configuration commands " \(page 42\)](#).

Other changes

For a detailed history of past releases of this document, see ["Revision history" \(page 10\)](#).

Revision history

September 2008

Standard 03.09. This document is up-issued to reflect changes in technical content.

July 2008

Standard 03.08. This document is up-issued to reflect changes in technical content.

June 2008

Standard 03.07. This document is up-issued to support SIP Firmware for IP Phone 1140E Release 2.0.

February 2008

Standard 02.04. This document is up-issued to include editing changes.

November 2007

Standard 02.03. This document is up-issued to reflect changes in the technical information in response to change requests. A note was added stating that NAT_SIGNALLING is required for networks that use STUN or SIP_PING for NAT traversal. The commands FORCE_OCT_END and OCT_END were changed to FORCE_OCT_ENDDIAL and DISABLE_OCT_ENDDIAL respectively.

October 2007

Standard 02.02. This document is up-issued to ensure that the value of 802.1P_MEDIA parameter in the QoS and ToS commands must be set to -1 to be able to have a voice path.

July 2007

Standard 02.01. This document is up-issued to support SIP Firmware for IP Phone 1140E Maintenance Release 1.1.

March 2007

Standard 01.02. This document is up-issued to incorporate an addition to the Maintenance and Troubleshooting section.

January 2007

Standard 01.01. This is a new document created to support the IP Phone 1140E with SIP Firmware.

How to Get Help

This section explains how to get help for Nortel products and services.

Getting Help from the Nortel Web site

The best way to technical support for Nortel products is from the Nortel Technical support Web site:

www.nortel.com/support

This site provides quick access to software, documentation, bulletins, and tools to address issues with Nortel products. More specifically, the site enables you to:

- download software, documentation, and product bulletins
- search the Technical Support Web site and the Nortel Knowledge Base for answers to technical issues
- sign up for automatic notification of new software and documentation for Nortel equipment
- open and manage technical support cases

Getting Help over the telephone from a Nortel Solutions Center

If you do not find the information you require on the Nortel Technical Support Web site, and have a Nortel support contract, you can also get help over the telephone from a Nortel Solutions Center.

In North America, call 1-800-4Nortel (1-800-466-7835).

Outside North America, go to the following Web site to obtain the telephone number for your region:

www.nortel.com/callus

Getting Help from a specialist using an Express Routing Code

To access some Nortel Technical Solutions Centers, you can use an Express Routing Code (ERC) to quickly route your call to a specialist in your Nortel product or service. To locate the ERC for your product or service, go to www.nortel.com/erc

Getting Help through a Nortel distributor or reseller

If you purchased a service contract for your Nortel product from a distributor or reseller, contact the technical support staff for that distributor or reseller.

Introduction to this guide

Subject

SIP Firmware Release 2.0 for IP Phone 1140E Administration (NN43113-300) describes how to install, configure, and provision the IP Phone 1140E for use on a SIP network.

Intended audience

This administration guide is intended for system administrators of the Nortel IP Phone 1140E with a basic understanding of SIP. This guide is not intended for end users of the IP Phone 1140E. Many of the tasks outlined in the guide influence the function of the IP Phone 1140E on the network and require an understanding of telephony and Internet Protocol (IP) networking.

Acronyms

This guide uses the following acronyms:

Table 1
Acronyms used

AAA	Authentication, Authorization, and Accounting
ALG	Application Layer Gateway
DHCP	Dynamic Host Configuration Protocol
DNS	Domain Name System
DRegex	Digit Regular Expression
DSCP	Differentiated Services Code Point
EAP	Extensible Authentication Protocol
ERE	Extended Regular Expressions
FQDN	Fully Qualified Domain Name
FTP	File Transfer Protocol
GARP	Gratuitous Address Resolution Protocol
GUI	Graphical User Interface
HTTP	Hyper Text Transfer Protocol

IETF	Internet Engineering Task Force
ISDN	Integrated Services Digital Network
IM	Instant Message
IP	Internet Protocol
IPCM	Internet Protocol Client Manager
ITU-T	Telecommunications Standardization sector of the International Telecommunications Union
LAN	Local Area Network
MAC	Media Access Control
MADN	Multiple Appearance Directory Number
MAS	Media Application Server
NAT	Network Address Translator
NetConfig	Configuration screens available after an IP Phone resets
NDU	Network Diagnostic Utility
PDT	Problem Determination Tool
PEC	Product Engineering Code
POE	Power Over Ethernet
POSIX	Portable Operating System Interface
PRACK	Provisional Acknowledgement
PSTN	Public Switched Telephone Network
PVQMon	Proactive Voice Quality Monitoring
QoS	Quality of Service
RADIUS	Remote Authentication Dial In User Service
RTCP	Real-time Control Protocol
RTCP XR	RTP Control Protocol Extended Reports
RTP	Real-time Transfer Protocol
SCA	Single Call Arrangement Shared Call Appearance
SDP	Session Description Protocol
SIMPLE	SIP for Instant Messaging and Presence Leveraging Extensions
SIP	Session Initiation Protocol
SMTP	Simple Mail Transfer Protocol
STUN	Simple Traversal of UDP through NAT devices
TFTP	Trivial File Transport Protocol
TPS	Terminal Proxy Server

TTL	Time-to-live
UDP	User Datagram Protocol
UFTP	UNISlim File Transfer Protocol
UNISlim	Unified Network IP Stimulus Protocol
VoIP	Voice over IP
VLAN ID	Virtual Local Area Network Identification
VLAN IP	Virtual Local Area Network Internet Protocol
VQMon	Voice Quality Monitoring

Related publications

Other publications related to the SIP Firmware Release 2.0 for IP Phone 1140E Administration are:

- *SIP Firmware Release 2.0 for IP Phone 1140E Quick Reference Card* (NN43113-102)
- *SIP Firmware Release 2.0 for IP Phone 1140E User Guide* (NN43113-101)
- *Broadsoft Partner Configuration Guide Nortel SIP Firmware for 1120E / 1140E IP Phones* (NN43121-300)
- *Expansion Module for IP Phones 1140E and 1120E (SIP Firmware) User Guide* (NN43110-301)
- Nortel IP Phone 1100 Series product bulletins on www.nortel.com/support.

Overview

Introduction

This chapter describes the hardware and firmware features of the Nortel IP Phone 1140E and provides a brief overview of Session Initiation Protocol (SIP).

SIP overview

Session Initiation Protocol (SIP) is a signaling protocol used for establishing multimedia sessions in an Internet Protocol (IP) network.

SIP is a text-based protocol similar to HTTP and SMTP. With the introduction of SIP to IP Phones, telephony integrates easily with other Internet services. SIP allows the convergence of voice and multimedia.

IP Phone 1140E with SIP Firmware

The IP Phone 1140E connects to an IP network using an Ethernet connection. All voice and signaling information is converted into IP packets and sent across the network.

The IP Phone 1140E can be ordered with UNISTim Firmware installed or with SIP Firmware installed. UNISTim Firmware and SIP Firmware use the same hardware, but the model number of an IP Phone 1140E with UNISTim Firmware is different from the model number of an IP Phone 1140E with SIP Firmware.

If you have an IP Phone 1140E with UNISTim Firmware, you can convert the firmware to SIP Firmware. To download the most recent version of SIP Firmware, see ["Download the SIP Firmware to the provisioning server"](#) (page 30).

This guide explains how to:

- configure the provisioning server and the DHCP server. Note: The provisioning server is where the firmware and the configuration files for the IP Phone 1140E reside. This is not the IP Client Manager (IPCM) of the call server.
- convert an IP Phone 1140E with UNISTim Firmware to an IP Phone 1140E with SIP Firmware

- provision the Device Settings parameters on the IP Phone 1140E with SIP Firmware

Note: Converting the firmware on an IP Phone 1140E from UNISTim Firmware to SIP Firmware overwrites the UNISTim Firmware. The IP Phone 1140E cannot operate in both modes simultaneously. A switch from UNISTim to SIP Firmware or SIP to UNISTim Firmware requires a firmware reload.

The following figure shows the main components of the IP Phone 1140E with SIP Firmware.

Figure 1
IP Phone 1140E with SIP Firmware



Related documentation

The *SIP Firmware Release 2.0 for IP Phone 1140E User Guide* (NN43113-101) tells the end user how to use the IP Phone 1140E, including how to:

- use the **Context-sensitive soft keys** and **Navigation key cluster**
- enter text
- use the address book
- access and use the call inbox and call outbox
- configure and use instant messaging
- receive, identify, answer, redirect, decline, or ignore an incoming call
- operate hold, three-way calling, call transfer, and call park
- use other features such as speed dial, call forward, do not disturb, and setting up conference calls

- configure Bluetooth headset operation (IP Phone 1140E only)

Note: All features are not available with all call servers.

The IP Phone 1140E Getting Started Card included in the box with the IP Phone 1140E also contains useful information and explains how to:

- connect the AC power adapter
- control the volume when answering a call
- make a call using the handset
- make a call with the headset or using handsfree
- use hold and mute
- set the contrast
- set the language

Installation overview

To install the IP Phone 1140E with SIP Firmware, three basic steps are required.

Procedure 1

Installation overview

Step	Action
1	Configure the provisioning server and, optionally, the DHCP server. The function of the provisioning server is to provide configuration options to every IP Phone 1140E throughout the network. The DHCP server can be configured to provide basic network-configuration data or a more comprehensive set of network-configuration data for the IP Phone 1140E with SIP Firmware.
2	Load SIP Firmware on the IP Phone 1140E.
3	Configure the initial network-configuration parameters on the IP Phone 1140E with SIP Firmware.

—End—

Figure 2
Installation of IP Phone 1140E with SIP Firmware, page 1 of 2

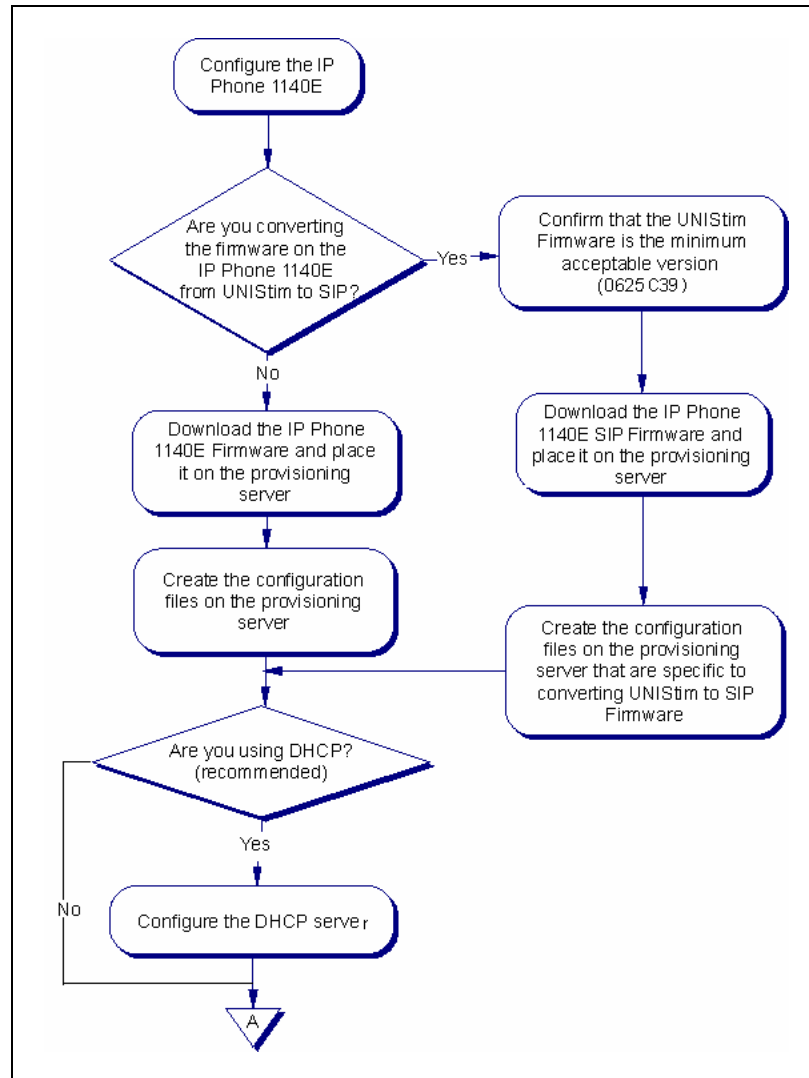
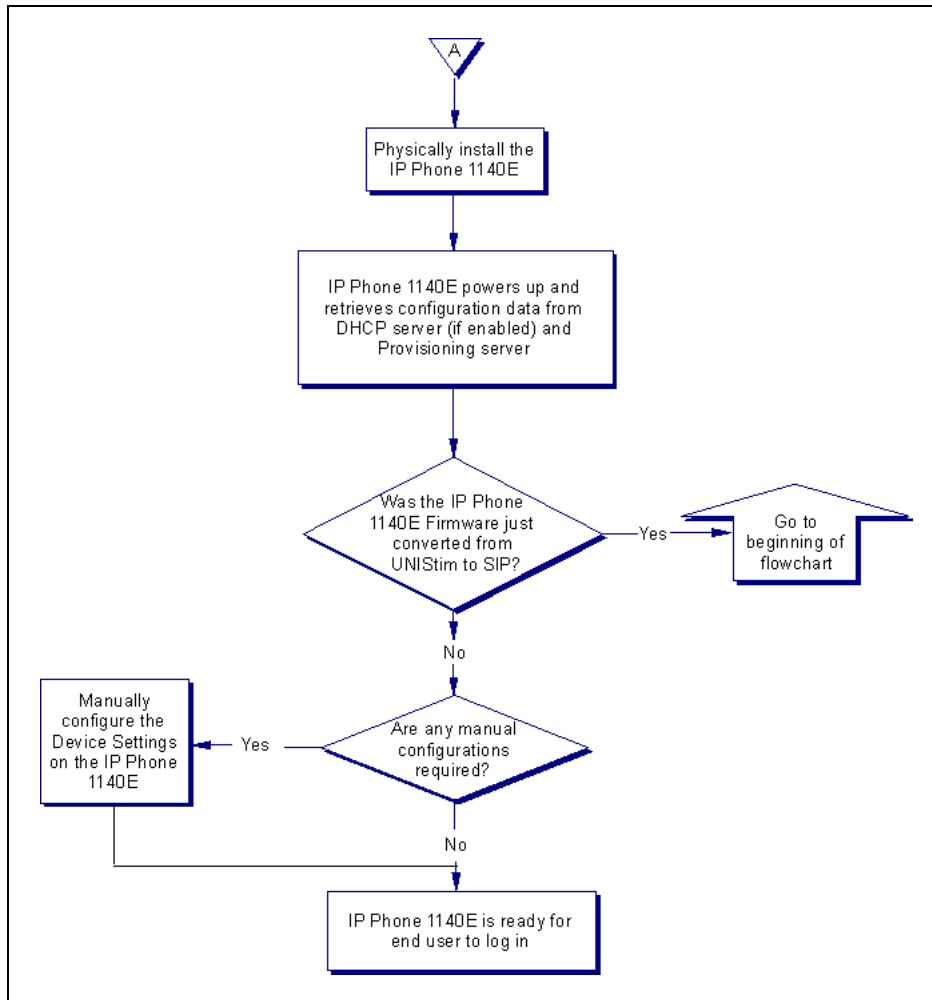


Figure 3
Installation of IP Phone 1140E with SIP Firmware, page 2 of 2



Supported SIP proxy servers

Call server	Call server release	Minimum SIP Firmware on the IP Phone 1140E
MCS 5100	3.5	1.0
MCS 5100	4.0	1.0
MCS 5100 with CS 1000	3.5 4.5	1.0
MCS 5100 with CS 1000	3.5 5.0	1.1
MCS 5100 with CS 1000	4.0 5.0	1.1
CS 1500	1.0	1.1
CS 2000	SN09U	1.1
CS 2000	SN10	1.1
CS 2100	SE09.1	1.1
CS 2100	SE10	1.1
BroadSoft	14.0	1.1

Before installation

Introduction

This chapter features a checklist of tasks you must complete before you install SIP Firmware on the IP Phone 1140E.

Preinstallation

Complete the following checklist.

Procedure 2

Preinstallation checklist

Step	Action
1	Read and become familiar with the <i>SIP Firmware Release 2.0 for IP Phone 1140E User Guide</i> (NN43113-101) and the <i>SIP Firmware Release 2.0 for IP Phone 1140E Quick Reference Card</i> (NN43113-102).
2	Ensure there is one IP Phone 1140E boxed package for each IP Phone 1140E being installed.
3	Ensure that the IP Phone 1140E box includes:

Table 2
IP Phone 1140E box contents

Item	Order number
IP Phone 1140E Graphite with icon keys without PS (SIP) (RoHS)	NTYS05CAE6 N0142353
IP Phone 1140E Graphite with English keys without PS (SIP) (RoHS)	NTYS05DAE6 N0142354
Accessories	
Handset, Charcoal	NTYS09AA70
Handset cord, Charcoal	NTYS10AA70

Footstand kit, Charcoal	NTYS11AA70
Telephone number label and lens kit	NTYS12AA
2.3 m (7 ft) CAT5 Ethernet cable	NTYS13AA

The IP Phone 1140E can be powered either by Power Over Ethernet (POE) or powered through an external power supply. Order the external power supply separately.

**WARNING**

Do not use the AC power adapter, if you are connected to a Power over the Ethernet (PoE) connection. Only use the AC power adapter when you do not have a Power over the Ethernet connection.

Table 3
1100 series phones separably orderable parts

CPC code	PEC code	Product description
N0146475	NTYS17BAE6	IP Phone Global Power Supply (2000, 1100, 1200) (RoHS)
N0089603	NTYS14AAE6	Standard IEC Cable - North America (RoHS)
A0781922	NTTK15AA	Standard IEC Cable – Australia / NZ (Note: RoHS not required)
N0114986	NTTK16ABE6	Standard IEC Cable – Europe
N0109787	NTTK17ABE6	Standard IEC Cable – Switzerland
N0109881	NTTK18ABE6	Standard IEC Cable – UK
N010978	NTTK22ABE6	Standard IEC Cable – Denmark
A0814961	A0814961	Standard IEC Cable - Argentina (Note: RoHS not required)
N0118951	NTTK26AAE6	Standard IEC Cable - Japan

**CAUTION**

The IP Phone 1140E must be plugged into a 10/100-BaseT Ethernet jack. Severe damage occurs if this IP Phone 1140E is plugged into an ISDN connection.

- 4 Ensure that the location meets the network requirements:
 - a DNS server and a DHCP server with DHCP relay agents installed, configured, and running. Using DHCP and DNS servers with CS 2000 network is recommended but not mandatory.
 - An Ethernet connection to a network with an appropriate SIP proxy server.

- One of the following file servers used as a Provisioning server:
 - TFTP server
 - FTP server
 - HTTP server

Only a TFTP server can be used for an initial UNISTim-to-SIP Firmware conversion. An IP Phone 1140E with SIP Firmware can operate with a TFTP, FTP, or HTTP file server.

—End—

Configure the provisioning server

ATTENTION

If you have UNISlim Firmware on your IP Phone 1140E, the firmware must be converted from UNISlim to SIP before you proceed with the following instructions. See the chapter ["Upgrade and convert the IP Phone 1140E firmware" \(page 79\)](#) for instructions on how to convert the firmware on an IP Phone 1140E from UNISlim to SIP.

If the IP Phone 1140E is installed with SIP Firmware, further SIP Firmware upgrades can be done with a TFTP, an FTP, or an HTTP server.

How provisioning works

Provisioning is performed without interaction with the call server. The IP Phone 1140E with SIP Firmware connects directly with the provisioning server in order to retrieve firmware files and configuration files. In this case, the provisioning server is not to be confused with the IP Client manager on the call server. The methods of provisioning are:

- Automatic provisioning at power-up: After the IP Phone 1140E powers up or is reset, it checks the provisioning server for the latest files.
- Provisioning through user interaction: The end user can manually check for updates by pressing the **Services** Context-sensitive soft key and selecting **SystemSystem**. In the menu select **Check for Updates**.
- Automatic provisioning at a preconfigured time: The IP Phone 1140E with SIP Firmware checks for updates every 24 hours, at a time specified by a parameter in the device configuration file.

The following steps are taken when provisioning updates occur. The IP Phone 1140E with SIP Firmware:

1. connects to the provisioning server
2. retrieves the provisioning file (1140eSIP.cfg) from the provisioning server
3. reads and acts upon the content of the provisioning file and decides whether any other file is needed, based on a set of rules. If files need to be downloaded to the IP Phone 1140E, a new file transfer session starts

for each file to be downloaded. The provisioning file (1140eSIP.cfg) can contain commands that prompt for confirmation before a file is downloaded.

Download the SIP Firmware to the provisioning server

To download the SIP Firmware, follow the steps in the next procedure.

Procedure 3

Downloading SIP Firmware for IP Phone 1140E from the Nortel Web site

Step	Action
------	--------

- | | |
|---|---|
| 1 | Go to www.nortel.com/support . |
| 2 | Log on to the Nortel Web site with a valid Nortel User ID and Password.

The Technical Support page appears. |
| 3 | Enter IP Phone 1140E in the Knowledge and Solution Engine box. |
| 4 | Select Software in the All types scroll down menu. |
| 5 | Press the gray arrow at the end of the Knowledge and Solution Engine box to obtain the Search Results . |
| 6 | From the Search Results , select the appropriate version of the SIP Firmware for the IP Phone 1140E, for example, SIP IP Phone 1140E Release 0625C39D26.bin . |
| 7 | Place the selected firmware on the provisioning server. |
-

—End—

Create the SIP provisioning file on the provisioning server

The provisioning file is downloaded from the provisioning server to the IP Phone 1140E every time the IP Phone 1140E checks for updates. The provisioning file is a clear text file that has the naming convention 1140eSIP.cfg. The following is an example of a provisioning file:

Figure 4
Sample provisioning file

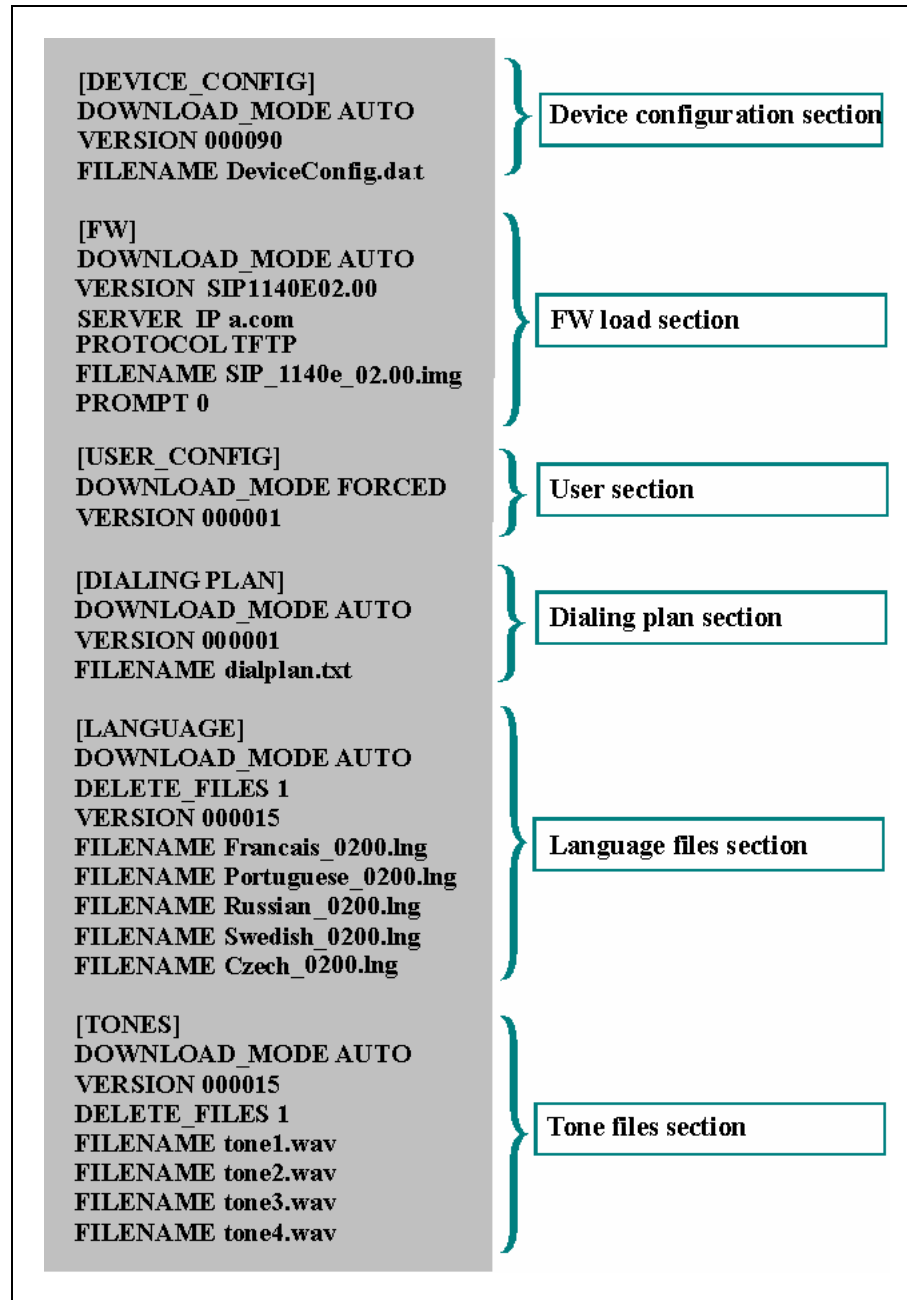


Table 4
Provisioning file supported sections

[DEVICE_CONFIG]	Device configuration file
[LANGUAGE]	Downloadable language files (more than one can be specified in each section)
[FW]	Firmware image
[DIALING_PLAN]	Dialing plan
[TONES]	Downloadable tones (.wav files)
[USER_CONFIG]	Set specific configuration file

Provisioning is performed using the commands in the 1140eSIP.cfg configuration file. The configuration file can have multiple sections.

Note: The maximum length of a line item in the configuration file is 80 characters. If a line item with more than 80 characters is encountered when parsing the configuration file, the remaining portion of the file following that line item will be ignored.

'#' is used to indicate a comment. Anything preceeding a '#' is a comment.

Each section in the configuration file defines rules for different file types. A section starts with a [SECTION NAME] to specify rules for each file type. For example: [FW].

A section is a mandatory field. Parsing of download rules for each file type starts with finding this key word. Currently, the following sections are supported by the IP Phone 1140E with SIP Firmware:

- **[FW]** section for the firmware load download
- **[DEVICE_CONFIG]** section for the device configuration
- **[DIALING_PLAN]** section for the dialing plan files
- **[LANGUAGE]** section for the language prompts files
- **[TONES]** section for the downloadable tone files
- **[USER_CONFIG]** section for set specific configuration file

Firmware [FW] image files originate from Nortel only and are authenticated during firmware download. If the FW authentication fails, the IP Phone 1140E displays an error message and continues operation with the existing FW image.

Device configuration files are used to set various parameters in the IP Phone 1140E.

Dialing plan files are used for configuring dialing patterns and the format of originated URIs in the SIP message.

Note: Some call servers do not yet support SIP URI dialing.

Language files are simple text files containing all text prompts used by the IP Phone 1140E. Language files are used for the localization of the IP Phone 1140E without firmware upgrade. Each language file has a header that contains a firmware load version with which this file is associated. Language files are signed by Nortel and are authenticated by the firmware for security reasons.

Tones files are standard in the Telecommunications Standardization sector of the International Telecommunications Union (ITU-T). The set supports custom tone files. The tone files must be WAV files with the following specification: A-law or u-law (8.0 kHz, 8-bit, mono or 16.0 kHz, 16 bit mono). The WAV files can be created and downloaded to the IP Phone 1140E. These files are not authenticated by the IP Phone 1140E.

Set specific configuration files support customizing the set on a per set/user level. Parameters in the device configuration file can be overwritten with a set specific configuration file.

Mandatory keywords in the Provisioning file are:

- **VERSION [xxxxxx]**, where xxxxxx is a six- to ten-digit number representing the version of the file on the server. The version of the module is specified in this field. The command is used for version comparison in AUTO mode. VERSION is mandatory for all sections. In the FW section, the firmware version of the load located on the provisioning server must be entered in this field. For all other sections, VERSION is just a counter that can be incremented if it is necessary to download a new file version.



CAUTION

The version number is stored permanently on the IP Phone 1140E until a higher version number is downloaded or until the version number is deleted using the **Srvcs, System, Erase User Data** menu selection on the IP Phone 1140E. This prevents a new file from being downloaded if a lower version number is used in the Provisioning File.

- **DOWNLOAD_MODE [AUTO | FORCED]** defines whether the version is checked. This command is optional. If this command is not present, AUTO mode is used as the default.
 - **AUTO** - This mode compares the version of the module from the VERSION field and the version of the module version stored in the FLASH memory of the IP Phone 1140E. The file download is initiated

only if the version specified is higher than the current version stored in the IP Phone 1140E. If the version is not applicable, as in the case of language files, the date of the file must be used for the decision.



CAUTION

The version number stored in the FLASH is permanent until a higher number is downloaded from the Provisioning file or you select **Srvcs, System, Erase User Data** on the IP Phone 1140E.

- **FORCED** - This mode forces the firmware download process. FORCED can be used for firmware downgrade procedures.

Note: In FORCED or AUTO_DOWNLOAD_MODE, the version number is overwritten with each firmware download.

- **FILENAME [filename]** specifies the file name to be downloaded for this section. For the language and tone section, the use of multiple filenames is allowed.

Optional keywords in the Provisioning file are:

- **PROMPT [YES | NO]** is used to indicate if the IP Phone 1140E should prompt the user for an update before the operation is performed. This command is optional with the default set to NO.
 - **YES** - enables the prompt
 - **NO** - disables the prompt
- **PROTOCOL [TFTP | FTP | HTTP]** defines the protocol used to download the file. The IP Phone 1140E with SIP Firmware supports TFTP, FTP and HTTP protocols for file download. This command is optional. If it is not present, the default protocol TFTP is used.

ATTENTION

When using the TFTP protocol to transfer the firmware image, the average round trip time must be < 75 ms. The IP Phone will timeout and abort the firmware download if the total download time is > 10 minutes.

If the average round trip time will be >75 ms, use the FTP or HTTP protocol to transfer the firmware image.

If using FTP or HTTP, **SRV_USER_NAME** and **SRV_USER_PASS** are also key words. These commands specify the credential used to login to the file server for file download. If not present, the protocol default

credential is used (no credential for TFTP and HTTP and anonymous with no password for FTP).

- **SERVER_IP [address]** allows the IP Phone 1140E to connect to the specified IP address or name of the server for which the file can be downloaded. If the address is not specified, the **SERVER_IP** that is used is the same **SERVER_IP** that is used to download the provisioning file.
- **DELETE_FILES [YES | NO]**, if present, erases the language and tone files stored in the IP Phone 1140E before new files are downloaded. Otherwise, new files with different names are added without erasing existing files. This command is optional. Note that there is a hard limit of 5 language files and 5 tone files that can be stored in the IP Phone 1140E. When the limits are exceeded, no new file can be accepted for download.
 - **YES** - erases the existing language and tone files
 - **NO** - does not erase existing language and tone files
- **SRV_USER_NAME [username]** - If the protocol is FTP or HTTP, this keyword specifies the user name to log on to the server.
- **SRV_USER_PASS [password]** - If the protocol is FTP or HTTP, this keyword specifies the password to log on to the server.

The downloading of these files is initiated when an IP Phone 1140E is powered on, when an automatic check for updates is invoked, or when you select **Srvcs, System, Erase User Data**. Any of these actions causes the IP Phone 1140E to contact the provisioning server and attempt to read the Provisioning file. A Soft Reset (**Srvcs, System, Reset Phone**) does not cause the IP Phone 1140E to retrieve the Provisioning file.

Setting the default language on the IP Phone 1140E to French

To set the default language on a new IP Phone 1140E, or an IP Phone 1140E that has not been logged into by an end user, include the following in the [DEVICE_CONFIG] and [LANGUAGE] sections of the 1140eSIP.cfg configuration file.

```
[DEVICE_CONFIG]
DOWNLOAD_MODE AUTO
VERSION 000002
FILENAME DeviceConfig.dat
```

```
[LANGUAGE]
DOWNLOAD_MODE AUTO
VERSION 0000000001
FILENAME French_d24.lng
```

The DeviceConfig.cfg file should contain the following.

```
DeviceConfig.cfg  
DEF_LANG French_d24
```

On a new IP Phone 1140E, the language will switch to French after downloading and processing the configuration files. The login menu will be in French. On a subsequent bootup, the login menu and all boot messages will be in French.

For a new user login, the set will create a new user profile. All menus remain in French. When a new user is created, the default language used is obtained from the DeviceConfig setting and stored as a user preference. After which the user preference for language is always used.

If a user has already logged in and either defaulted or chosen English as the user language preference, changing the configuration files will not affect the users language display.

Create the device configuration file on the provisioning server

After the IP Phone 1140E downloads the provisioning file, the IP Phone 1140E reads the [DEVICE_CONFIG] section and is directed to download the device configuration file.

The device configuration file is a clear text file and the naming convention is defined by the administrator. See the FILENAME keyword in the [DEVICE_CONFIG] section of the SIP provisioning file.

The following is an example of a device configuration file.

Figure 5
Sample device configuration file

```
# Server and Network configuration commands
DNS_DOMAIN corp.yourcompany.com
SIP_DOMAIN1 yourcompany.com
SERVER_IP1_1 10.1.2.3
SERVER_IP1_2 10.1.2.4
SERVER_PORT1_1 5060
SERVER_PORT1_2 5060
SERVER_RETRIES1 3
DEF_USER1 user1

# Voice Feature configuration commands
VMAIL 5555
VMAIL_DELAY 300

# Administrative feature commands
BANNER MCS_4.0
AUTOLOGIN_ENABLE YES

# Voice Application commands
DEF_LANG English
DEF_AUDIO_QUALITY High
ENABLE_BT YES
ENABLE_3WAY_CALL NO
```

The next table provides a summary of the commands that can be used in the device configuration file. The exact syntax of each command follows the table.

Table 5
Device configuration commands

Server and network configuration commands	SIP_DOMAIN1	SERVER_PORT3_2
	SIP_DOMAIN2	SERVER_PORT4_1
	SIP_DOMAIN3	SERVER_PORT4_2
	SIP_DOMAIN4	SERVER_PORT5_1
	SIP_DOMAIN5	SERVER_PORT5_2
	SERVER_IP1_1	SERVER_RETRIES1

	SERVER_IP1_2 SERVER_IP2_1 SERVER_IP2_2 SERVER_IP3_1 SERVER_IP3_2 SERVER_IP4_1 SERVER_IP4_2 SERVER_IP5_1 SERVER_IP5_2 SERVER_PORT1_1 SERVER_PORT1_2 SERVER_PORT2_1 SERVER_PORT2_2 SERVER_PORT3_1	SERVER_RETRIES2 SERVER_RETRIES3 SERVER_RETRIES4 SERVER_RETRIES5 DNS_DOMAIN DEF_USERS DEF_USER1 DEF_USER2 DEF_USER3 DEF_USER4 DEF_USER5 UPDATE_USERS SIP_PING
Feature configuration commands	VMAIL VMAIL_DELAY AUTOLOGIN_ENABLE AUTO_UPDATE AUTO_UPDATE_TIME AUTO_UPDATE_TIME_RANGE TRANSFER_TYPE REDIRECT_TYPE ENABLE_PRACK	MAX_INBOX_ENTRIES MAX_OUTBOX_ENTRIES MAX_REJECTREASONS MAX_CALLSUBJECT MAX_PRESENCENOTE DEF_LANG MAX_IM_ENTRIES MAX_ADDR_BOOK_ENTRIES ADDR_BOOK_MODE DEF_AUDIO_QUALITY
Feature configuration commands (continued)	PROXY_CHECKING ENABLE_BT AUTH_METHOD BANNER FORCE_BANNER DST_ENABLED TIMEZONE_OFFSET FORCE_TIME_ZONE IM_MODE	HOLD_TYPE ENABLE_3WAY_CALL DISABLE_PRIVACY_UI DISABLE_OCT_ENDDIAL FORCE_OCT_ENDDIAL SNTP_ENABLE SNTP_SERVER MADN_TIMER MADN_DIALOG

	IM_NOTIFY DEF_DISPLAY_IM CALL_WAITING DISTINCTIVE_RINGING USE_RPORT TOVM_SOFTKEY_ENABLE TOVM_VOICEMAIL_ALIAS TOVM_VOICEMAIL_PARAM MAX_RING_TIME ENABLE_UPDATE	DEFAULT_CFWD_NOTIFY FORCE_CFWD_NOTIFY RTP_MIN_PORT RTP_MAX_PORT SCA_HOLD_BEHAVIOR SCA_APPEARANCES SCA_BROADWORKS EXP_MODULE_ENABLE FORCE_REBOOT PROMPT_ON_LOCATION_OTHER
QoS and ToS commands	DSCP_CONTROL 802.1P_CONTROL DSCP_MEDIA	802.1P_MEDIA DSCP_DATA 802.1P_DATA
Tone configuration commands	DIAL_TONE RINGING_TONE BUSY_TONE	FASTBUSY_TONE CONGESTION_TONE
NAT configuration commands	NAT_SIGNALLING NAT_MEDIA NAT_TTL	STUN_SERVER_IP1 STUN_SERVER_IP2 STUN_SERVER_PORT1 STUN_SERVER_PORT2
Voice Quality Monitoring (VQMon) configuration commands	VQMON_PUBLISH VQMON_PUBLISH_IP LISTENING_R_ENABLE LISTENING_R_WARN LISTENING_R_EXCE PACKET_LOSS_ENABLE PACKET_LOSS_WARN	PACKET_LOSS_EXCE JITTER_ENABLE JITTER_WARN JITTER_EXCE DELAY_ENABLE DELAY_WARN DELAY_EXCE SESSION_RPT_EN SESSION_RPT_INT
System commands	ADMIN_PASSWORD	

Device configuration commands



CAUTION

The syntax of the device configuration file is case sensitive. Verify that the commands entered follow the case defined in this document.

ATTENTION

Parameters in the device configuration file with empty values are not allowed and will cause write failure.

Server and network configuration commands

- **SIP_DOMAIN[x] [domain_name]** preconfigures the proxy domain name for all servers. The same configuration can be done through the domain configuration menu on the IP Phone 1140E.
 - **x** - the number of the SIP domain number from 1 to 5.
 - **domain_name** - the proxy domain name for all servers.

Note: SIP_DOMAIN[x] is provisioned after user logout.
- **SERVER_IP[x] [y] ip_address** configures the primary and secondary IP address for each domain, two proxies for each domain.
 - **x** - the domain number from 1 to 5.
 - **y** - the corresponding primary and secondary IP addresses. y=1 indicates the primary address and y=2 indicates the secondary address.
 - **ip_address** - the IP address of the SIP proxy server.
- **SERVER_PORT[x] [y] [port_number]** configures the signaling ports for each proxy.
 - **x** - the domain number.
 - **y** - the corresponding primary and secondary IP addresses. y=1 indicates the primary address and y=2 indicates the secondary address.
 - **port_number** - the SIP proxy signaling port (default is 5060).
- **SERVER_RETRIES[x] [number_of_retries]** confirms the number of retries for each domain. The default number of retries is 3.
 - **x** - the domain number from 1 to 5.

- **number_of_retries** - the number of retry attempts to connect to the proxy server.
- **DNS_DOMAIN [domain]** is the DNS domain of the IP Phone 1140E.
- **DEF_USERS[x] [user_name]** allows you to enter the default user name for all domains. When the device configuration file gets downloaded, the default user name is used when logging in.
 - **x** - the domain number from 1 to 5.
 - **user_name** - the default user name.
- **UPDATE_USERS [YES | NO]** affects the default user names stored in the IP Phone 1140E. If this flag is set to YES, the default user names are overwritten each time a new device configuration file is downloaded.
 - **YES** - the default user names are overwritten each time a new device configuration file is downloaded.
 - **NO** - the default user names are not overwritten each time a new device configuration file is downloaded.
- **SIP_PING [YES | NO]** The SIP_PING configuration value is used to maintain server heartbeat detection and to keep a firewall pinhole open.

When used for server heartbeat detection, the IP Phone 1140E periodically pings the SIP Proxy and awaits a response. When three attempts to ping the SIP Proxy fail, the IP Phone 1140E begins a failover process and attempts to connect to the next configured SIP Proxy IP in the same domain.

When a NAT TRAVERSAL method is selected, the SIP_PING configuration value also helps keep a firewall pinhole open.

ATTENTION

Decide carefully whether SIP_PING usage is appropriate for your environment. Even when SIP_PING is not used for NAT TRAVERSAL, it is highly likely that you must keep SIP_PING enabled for server heartbeat detection.

If the IP Phone 1140E is behind a firewall, it is very likely that you must keep SIP_PING enabled, unless an alternate method of keeping the firewall pinhole open is used.

The default value is yes if not specified in the device configuration file. If SIP_PING is changed in the Device configuration file, the IP Phone 1140E must be rebooted for the change to take effect.

- **YES** - enables ping

- **NO** - disables pinging

Feature configuration commands

- **TOVM_SOFTKEY_ENABLE [YES | NO]**

The transfer to Voice Mail feature is supported on MCS 5100 servers.

- **YES** - enables the toVM soft key on the phone.
- **NO** - disables the toVM soft key on the phone.

- **TOVM_VOICEMAIL_ALIAS <string>** customizes the user ID of the SIP URI of the voice mail system. The transfer to Voice Mail feature is supported on MCS 5100 servers.
- **TOVM_VOICEMAIL_PARAM<string>** customizes the parameter name of the SIP URI of the voice mail system. The transfer to Voice Mail feature is supported on MCS 5100 servers.
- **SCA_APPEARANCES** sets the maximum number of appearances that will be used for outgoing calls by the Shared Call Appearance (SCA) group. The valid range for this parameter is 2 to 24. The default value is 12. The SCA feature is supported on BroadWorks servers.
- **SCA_BROADWORKS [YES | NO]**

The SCA feature is supported on BroadWorks servers.

 - **YES** - activates the Broadsoft Shared Call Appearance (SCA) feature.
 - **NO** - deactivates the Broadsoft Shared Call Appearance (SCA) feature. This is the default option.
- **SCA_HOLD_BEHAVIOR [PRIVATE | PUBLIC]** sets the default behavior of the hold button when user determined behavior does not exist. When a user creates a new profile the default behavior is taken from this setting. After the creation of a new profile this configuration setting is not used. The default option is PUBLIC. The SCA feature is supported on BroadWorks servers.
- **RTP_MIN_PORT**

The minimum RTP port value is an integer between 1024 and 65535, exclusive of the restricted SIP ports between 5059 and 5080. The default value is 50000.
- **RTP_MAX_PORT**

The maximum RTP port value is an integer between 1024 and 65535, exclusive of the restricted SIP ports between 5059 and 5080. The default value is 50100.

Note 1: The RTP port configuration parameters must satisfy the constraints that (RTP_MAX_PORT - RTP_MIN_PORT) is greater than or equal to 100 and less than 1000.

Note 2: If there is a provisioning error, RTP_MIN_PORT is reset to the default value of 50000 and RTP_MAX_PORT is reset to the default value of 50100. An error message will be logged. The SystemConfig file stores 50000 and 50100, rather than the erroneous configuration values, to indicate that the configuration attempt has been rejected.

- **CALL_WAITING [SPEAKER | STREAM]**
 - **SPEAKER** - the call waiting tone is played on the phone speaker. This is the default option.
 - **STREAM** - the call waiting tone is injected into the stream played on the transducer in use for the active call
- **DISTINCTIVE_RINGING [YES | NO]**
 - **YES** - turns on the distinctive ringing feature. This is the default option.
 - **NO** - turns off the distinctive ringing feature.
- **USE_RPORT [YES | NO]**
 - **YES** - allows the phone to work from behind and/or in front of a symmetrical NAT with servers and/or clients that support RFC 3581.
 - **NO** - disables implementation of support for RFC 3581. This is the default option.

Note: To provision USE_RPORT, the IP Phone 1140 must be rebooted after the device configuration file is updated. To force a hard reboot after the device configuration file is updated set FORCE_REBOOT YES.
- **EXP_MODULE_ENABLE [YES | NO]**
 - **YES** - the set will detect and enable an expansion module.
 - **NO** - the set will not detect an expansion module. This is the default option.
- **MAX_RING_TIME [x]** - an integer between 30 and 600 that sets the number of seconds for incoming calls to ring before ignoring them. The default value is 120.
- **ENABLE_UPDATE [YES | NO]**

- **YES** - enables UPDATE message support and adds "UPDATE" to ALLOW header. This is the default option.
- **NO** - disables UPDATE message support.

Note: ENABLE_UPDATE is provisioned after user logout.

- **FORCE_REBOOT [YES | NO]**

- **YES** - forces hard reboot after device configuration update.
- **NO** - does not force hard reboot after device configuration update. This is the default option.

Note: In order for FORCE_REBOOT to reboot the phone the VERSION of the device configuration file must be incremented, even if DOWNLOAD_MODE is set to FORCED.

- **PROMPT_ON_LOCATION_OTHER [YES | NO]**

- **YES** - prompt the user to select new location if location "other" was previously selected.
- **NO** - do not prompt the user to select new location if location "other" was previously selected. This is the default option.

- **VMAIL [vmail_number]** is the voice mail address which can be the URI or the DN number of the voice mail server. This command takes a string as a parameter. This is the default link for a new user profile only. Individual users can customize the link through **Prefs, Message Options, Voice Mail Settings**. This command has no effect on the user profiles after it is created.

- **vmail_number** - the number or URI of the voicemail server.

- **VMAIL_DELAY[x]** is a delay, configured in milliseconds, between when the voice mail server answers the call and the start of dialing the voice mail user ID. The default value is 1000ms.

- **x** - the delay in milliseconds

- **AUTOLOGIN_ENABLE [YES | NO | USE_AUTOLOGIN_ID] or [1 | 0 | 2]** controls whether the set attempts to automatically login to the proxy server.

- **YES** - turns on the auto login feature.
- **NO** - turns off the auto login feature.
- **USE_AUTOLOGIN_ID** - enables the auto login id feature using the userid specified in AUTOLOGIN_ID_KEY01 and the password

specified in AUTOLOGIN_PASSWD_KEY01 to register and authenticate. Both userid and password must be specified.

The AUTOLOGIN_ID_KEY01 and AUTOLOGIN_PASSWD_KEY01 parameters are defined in the set specific configuration file.

Note: When using this setting, the user will be prevented from logging out of the phone.

or

- **1** - turns on the auto login feature.
- **0** - turns off the auto login feature.
- **2** - enables the auto login id feature using the userid specified in AUTOLOGIN_ID_KEY01 and the password specified in AUTOLOGIN_PASSWD_KEY01 to register and authenticate. Both userid and password must be specified.

The AUTOLOGIN_ID_KEY01 and AUTOLOGIN_PASSWD_KEY01 parameters are defined in the set specific configuration file.

Note: When using this setting, the user will be prevented from logging out of the phone.

Note: If auto login id is enabled in the set specific configuration file, it is recommended that AUTOLOGIN_ENABLE be set to either Yes/No or 1/0 in the device configuration file. This recommendation facilitates migrating a set that uses the set specific configuration file to not using the set specific configuration file. The migration to just using the device configuration file can be done by deleting the set specific configuration file. If the device configuration file does not have the matching parameters in the set specific configuration file, the set will continue to use the previously assigned settings after the set specific configuration file is deleted. This recommendation applies to other parameters in the set specific configuration file.

- **AUTO_UPDATE [YES | NO]** is a command to enable or disable the automatic updating of the IP Phone 1140E with SIP Firmware configuration files from the provisioning server. Enabling this command causes the IP Phone 1140E with SIP Firmware to check for updates once every day. The default is disabled.
 - **YES** - turns on the AUTO_UPDATE feature.
 - **NO** - turns off the AUTO_UPDATE feature.
- **AUTO_UPDATE_TIME [x]** is the actual time in seconds, starting from midnight, before an automatic update occurs. Each IP Phone 1140E adds random numbers to the time specified by this command so every

IP Phone 1140E does not try to access the provisioning server at the same time. By default the automatic update feature is disabled (see `AUTO_UPDATE_RANGE`).

- **x** - the time after midnight that the automatic update occurs.
- **AUTO_UPDATE_TIME_RANGE [x]** is the range in hours, from the `AUTO_UPDATE_TIME` whereby an IP Phone 1140E checks for updates from the server. The default range is 1 hour.
 - **x** - the range in hours when the IP Phone checks for updates from the server. The range can be from 1 to 6 hours.
- **TRANSFER_TYPE [MCS | STANDARD]** is used to configure the IP Phone 1140E to activate Nortel conference server-assisted attended transfers, instead of the industry standard method of attended transfers. The default setting is MCS.
 - **MCS** - the typical attended transfer used by Nortel proxies. MCS uses a conference server to do the attended transfer.
 - **STANDARD** - the standard method of a transfer. This method does not involve a conference server.
- **REDIRECT_TYPE [MCS | RFC3261]** is a command used to select different protocols for set redirection. The default setting is MCS.
 - **MCS** - when the IP Phone 1140E receives either 301 (moved permanently) or 302 (moved temporarily) during registration, it is assumed the IP Phone 1140E is moved to a new MCS 5100 system (proxy+registrar) and all subsequent messages are sent to the new address.
 - **RFC3261** - the IP Phone 1140E assumes that, if during registration, a 301 (moved permanently) is received, the message contains a new registrar address. The IP Phone 1140E tries to register to the registrar using the existing proxy.
- **ENABLE_PRACK [YES | NO]** PRACK is utilized to make some SIP messages reliable and requires that an ACK be sent with many SIP messages. `ENABLE_PRACK` is often utilized to verify that early media is being received. See RFC 3262 for details.
 - Note 1:** `ENABLE_PRACK` must be set to NO when connected to the MCS 5100 Release 3.5 system.
 - Note 2:** `ENABLE_PRACK` is provisioned after user logout.
 - **NO** - disables PRACK and is the default value.
 - **YES** - enables PRACK.

- **PROXY_CHECKING [YES | NO]** enables and disables extra security checking when incoming requests are sent to the IP Phone 1140E. The IP Phone 1140E with SIP Firmware always sends requests through an outgoing proxy. However, it is possible, through this configuration, to be able to accept an incoming request directly or through an incoming proxy.
 - **YES** - means that the request must come directly from the proxy server. YES is the default to enable proxy checking.
 - **NO** - means the request can be sent directly to the IP Phone 1140E. (NO is only suitable in a few situations).
- **ENABLE_BT [YES | NO]** is a flag to enable and disable Bluetooth support in the IP Phone 1140E.
 - **YES** - enables Bluetooth.
 - **NO** - disables Bluetooth. The default is NO.

Note that the IP Phone 1140E hardware does not support Bluetooth and this command is ignored.
- **DEF_AUDIO_QUALITY [Low | Medium | High]** is a command used for setting the default audio quality used for each new call. Audio quality can be changed when the call is active. If this command is not present in the configuration file, the IP Phone 1140E uses High quality as its default value. The possible parameters for this command are High, Medium, and Low. If any other parameter is entered or these commands are misspelled, the IP Phone 1140E uses High as the default setting.

The following codecs are used for each selection:

 - **Low** - G729 ptime 30.
 - **Medium** - G711 ptime 30.
 - **High** - G711 ptime 20.
- **AUTH_METHOD [AUTH | AUTH_INT]** is used to configure the SIP authentication method.
 - **AUTH** - only authenticates (username/password)
 - **AUTH_INT** - authentication plus integrity checking (an MD5 hash of the entity is also computed and checked).
- **BANNER [banner_text]** preconfigures the banner on the IP Phone 1140E. Use a text string to set the banner. For example, BANNER ABC Company sets the banner to ABC Company. The text string can have a maximum of 24 characters.
 - **banner_text** - an ASCII string displayed on the screen of the IP Phone 1140E with SIP Firmware.

- **FORCE_BANNER [YES | NO]** is set by the system administrator through the configuration file. If FORCE_BANNER is set to YES, the banner from the configuration file is reloaded each time the IP Phone 1140E powers up, even if the user changes the banner manually.
 - **YES** - causes the banner set by the administrator to override any banner set by the user.
 - **NO** - allows the user to set the banner.
- **DST_ENABLED [YES | NO]** enables and disables the daylight saving time mechanism. The time received from the server is GMT and is converted to the proper timezone by the IP Phone 1140E. If the Daylight Saving Time feature is enabled, the IP Phone 1140E automatically calculates the DST time at the appropriate date and converts the time to and from DST. The calculations used are based on the new rules applicable to DST in 2007. The IP Phone 1140E is set up to use the North American DST scheme.
 - **YES** - enables daylight savings time.
 - **NO** - disables daylight savings time.
- **TIMEZONE_OFFSET [x]** is used to set the current time zone offset from GMT in seconds. TIMEZONE_OFFSET takes a number as a parameter. For example, TIMEZONE_OFFSET -25200 sets the time zone offset to MST, which is GMT-7 ($-7 \times 3600 = -25200$ seconds).

Table 6
Time zone offset

Location	Time zone offset (seconds)
(GMT-10:00) Hawaii	-36000
(GMT-09:00) Alaska	-32400
(GMT-08:00) Pacific time (US and Canada)	-28800
(GMT-07:00) Mountain time (US and Canada)	-25200
(GMT-06:00) Central time (US and Canada)	-21600
(GMT-05:00) Eastern time (US and Canada)	-18000
(GMT-04:00) Atlantic time (US and Canada)	-14400
(GMT-03:00) Brasilia, Buenos Aires	-10800
(GMT+00:00) Greenwich, Dublin, Lisbon, London	0
(GMT+01:00) Amsterdam, Berlin, Rome, Stockholm, Madrid, Paris	3600
(GMT+02:00) Athens, Istanbul, Cairo, Helsinki, Jerusalem	7200

(GMT+03:00) Moscow, St. Petersburg	10800
(GMT+05:30) Bombay, Calcutta, Madras, New Delhi	18000
(GMT+08:00) Beijing, Chongqing, Hong Kong, Singapore, Taipei	28800
(GMT+09:00) Osaka, Sapporo, Tokyo, Seoul	32400
(GMT+10:00) Canberra, Melbourne, Sydney	36000
(GMT+12:00) Auckland, Wellington	43200

- **FORCE_TIME_ZONE [YES | NO]** allows you to force the timezone offset on each user's IP Phone 1140E. The default is NO.
 - **YES** - forces the IP Phone to use the TIMEZONE_OFFSET specified in the device configuration file.
 - **NO** - uses the value stored in the user preferences.
- **IM_MODE [ENCRYPTED | TEXT | SIMPLE | DISABLED]** is used to configure the mode of instant messaging (IM). The default setting is ENCRYPTED.
 - **ENCRYPTED** - instant messages are sent encrypted.
 - **TEXT** - instant messages are sent as text.
 - **SIMPLE** - instant messages are sent using SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE) protocol.
 - **DISABLED** - instant messaging is turned off and no instant messages can sent or received.
- **IM_NOTIFY [YES | NO]** is used to turn on or off the Blue LED indicator upon receipt of an instant message.
 - **YES** - the Blue LED functions when an instant message is received.
 - **NO** - the Blue LED does not function when an instant message is received.

Note: If IM_NOTIFY is disabled, the Blue LED continues to operate for other features.
- **DEF_DISPLAY_IM [YES | NO]** enables or disables the display of instant messages (IM). The default setting is NO.
 - **YES** - enables display of IMs.
 - **NO** - disables display of IMs.
- **MAX_INBOX_ENTRIES [x]** is used to restrict the maximum number of inbox entries and takes a number as a parameter. For example,

MAX_INBOX_ENTRIES 100 limits the number of entries in the inbox to 100. The default limit is 100.

— **x** - the maximum number of in box entries.

- **MAX_OUTBOX_ENTRIES [x]** is used to restrict the maximum number of outbox entries and takes a number as a parameter. For example, **MAX_OUTBOX_ENTRIES 100** limits the number of entries in the outbox to 100. The default limit is 100.

— **x** - the maximum number of outbox entries.

- **MAX_REJECTREASONS [x]** is used to restrict the maximum number of Call Decline Reasons (**Prefs, Feature Options, Call Decline Reasons**) and takes a number as a parameter. The default limit is 20.

— **x** - the maximum number of reject reasons.

- **MAX_CALLSUBJECT [x]** is used to restrict the maximum number of call subjects (**Prefs, Feature Options, Call Subject**) and takes a number as a parameter. The default limit is 20.

— **x** - the maximum number of call subject reasons.

- **MAX_PRESENCENOTE [x]** is used to restrict the maximum number of presence notes and takes a number as a parameter. The default limit is 20.

— **x** - the maximum number of presence notes that an IP Phone 1140E can receive.

- **DEF_LANG [language]** is a command used for setting the default language. Select one of the supported languages from the language list downloaded. Note that the corresponding language file must be downloaded and stored in the IP Phone 1140E through the [LANGUAGE] section in Provisioning. If the language file is not stored in the IP Phone 1140E, the default language English is used.

— **language** - the default language used.

- **MAX_IM_ENTRIES [x]** is used to set the maximum number of instant message (IM) entries and takes a number as a parameter. Once the maximum number is reached, the oldest IM is deleted without any user notification. The default limit is 999.

— **x** - the maximum number of instant messages.

- **MAX_ADDR_BOOK_ENTRIES [x]** is used to set the maximum number of entries in the address book and takes a number as a parameter. The default limit is 100.

- **x** - the maximum number of address book entries.
- **ADDR_BOOK_MODE [NETWORK | LOCAL | BOTH]** is a command to choose the address book that is used to search for other users. The default setting is NETWORK.
 - **NETWORK** - downloads the user's address book from the network (MCS). New address book entries are uploaded to the network.
 - **LOCAL** - creates a user address book and stores it locally on the IP Phone
 - **BOTH** - attempts to download a network address book and keep a copy on the IP Phone. If a network address book is available, the IP Phone 1140E functions as if NETWORK mode has been selected.
- **HOLD_TYPE [RFC2543 | RFC3261]** is used to select the protocol to hold a call. The default setting is RFC3261.
 - **RFC2543** - RFC 2543 is a standard protocol of the Internet Engineering Task Force (IETF).
 - **RFC3261** - RFC3261 is a standard protocol of the IETF.
- **ENABLE_3WAY_CALL [YES | NO]** is a flag to enable or disable local phone-based three-way calling for three-party conferences.
 - **YES** - enables local (phone-based) three-way calling for three-party conferences. YES is the default.
 - **NO** - disables local (phone-based) three-way calling.
- **DISABLE_PRIVACY_UI [YES | NO]** is a flag to disable the privacy setting in UI menus. Disabling the privacy setting in UI menus disables the user ability to set privacy options.
 - **YES** - disables the privacy setting in the UI menus.
 - **NO** - enables the privacy setting in the UI menus. NO is the default.
- **DISABLE_OCT_ENDDIAL [YES | NO]** is a flag used to configure the pound (#) key. The default setting is YES.
 - **YES** - the pound (#) key initiates dialing when pressed after a telephone number is entered.
 - **NO** - the pound (#) key functions as any other digit or character on the dial pad typically used in networks that use vertical service codes or access codes.

- **FORCE_OCT_ENDDIAL [YES | NO]** is a flag used to override attempts to change the function of the pound (#) key on the Graphical User Interface (GUI). The default setting is NO.
 - **YES** - overrides attempts to change the function of the pound (#) key on the GUI.
 - **NO** - does not override a change of the function of the pound (#) key on the GUI.

- **SNTP_ENABLE [YES | NO]** allows the IP Phone 1140E to obtain the time and date from an NTP server. The default is NO.

The IP Phone 1140E updates the time once every 24 hours from the NTP server. If the IP Phone 1140E cannot contact the server, the IP Phone 1140E tries every 15 minutes up to a maximum of 6 attempts, and then hourly attempts are made. If **SNTP_ENABLE** is set to NO the IP Phone 1140E tries to retrieve the time and date from the SIP proxy server. However, not all SIP proxy servers support this method of retrieving the time and date.

- **YES** - enables NTP.
- **NO** - disables NTP.

- **SNTP_SERVER [ip_address]** is the IP address or FQDN of the NTP server that provides the time and date to the IP Phone 1140E. If this is not specified, the IP Phone does not generate any NTP requests.

- **ip_address** - the IP address of the NTP server in either Fully Qualified Domain Name (FQDN) or non-FQDN format.

- **MADN_TIMER [x]** is used to set the MADN polling timer interval (the interval at which the IP Phone 1140E attempts to determine the MADN group of the logged-in user). The minimum value for the polling interval is 900 seconds (15 minutes). The default value is 1800.

- **x** - the time delay (in seconds) between queries to find the MADN group DN of a user. The minimum value 900.

- **MADN_DIALOG [YES | NO]** is used to set the SIP URI or the GROUP DN for the subscription to the dialog event. The default value is NO.

- **YES** - subscribes to the dialog event using the SIP URI of the user.
- **NO** - subscribes to the dialog event using the group of the user.

Note: The SIP URI is used to subscribe for MADN support for CS 2000.

- **DEFAULT_CFWD_NOTIFY [YES | NO]** is used to configure the "ring splash" which occurs when either local call forwarding (MCS 5100) or network-based call forwarding have been enabled (CS 2000, CS 2100). If this configuration value is enabled, the IP Phone 1140E plays an abbreviated ring tone to remind the user that a call has been forwarded. This configuration value only effects users when their user profile is first created, unless the FORCE_CFWD_NOTIFY flag is also used. The default setting is NO
 - **YES** - a brief ring splash plays when a call is forwarded.
 - **NO** - the ring splash is not be played.
- **FORCE_CFWD_NOTIFY [YES | NO]** allows the administrator to force the behavior of the DEFAULT_CFWD_NOTIFY value on all users who login to the IP Phone 1140E. The default setting is NO.
 - **YES** - the DEFAULT_CFWD_NOTIFY configuration value is forced into effect for the user.
 - **NO** - the configuration value is not be forced into effect for the user.

QoS and ToS commands

- **DSCP_CONTROL [x]** is a value entered in decimal format between -1 and 63. If the value is -1, the DSCP value is picked up by the Service Package. The default value is 40.
 - **x** - a value from -1 to 63 indicating the DSCP value.
- **802.1P_CONTROL [x]** is a value entered in decimal format between -1 and 7 representing the 802.1P value in the SIP signaling packets. If the value is -1, the 802.1P value is retrieved from the Service Package. The default value is 6.
 - **x** - the value from -1 to 7 indicating the 802.1P value.
- **DSCP_MEDIA [x]** is a value entered in decimal format between -1 and 63 representing the DSCP value in the Real-time Transfer Protocol packets. If the value is -1, the DSCP value is retrieved from the Service Package. The default value is 44.
 - **x** - a value from -1 to 63 indicating the DSCP value.
- **802.1P_MEDIA [x]** is a value entered in decimal format between -1 and 7 representing the 802.1P value in the IP Phone 1140 Media (RTP) packets. If the value is -1 then the 802.1P value is retrieved from the Service Package is the 802.1 setting for media Real-time Transport Protocol (RTP). The default value is -1.
 - **x** - a value from -1 to 7 indicating the 802.1P value.

- **DSCP_DATA [x]** is a value entered in decimal format between -1 and 63 representing the DSCP value in the provisioning packets. If the value is -1, the DSCP value is retrieved from the Service Package. The default value is 40.
 - **x** - a value from -1 to 63 indicating the DSCP value.
- **802.1P_DATA [x]** is a value entered in decimal format between -1 and 7 representing the 802.1P value in the provisioning packets. If the value is -1, the 802.1P value is retrieved from the Service Package. The default value is 6.
 - **x** - a value from -1 to 7 indicating the 802.1P value.

Tone configuration commands

- **DIAL_TONE [frequency1 | frequency2 | on_time | off_time]** is used to select the tone advising the caller that the exchange is ready to receive call information and invites the user to start sending call information. You can select the country specific tone. The default tone is the North American tone.
 - **frequency1** - the frequency of tone 1.
 - **frequency2** - the frequency of tone 2.
 - **on_time** - the duration of the tone when it is on. A -1 indicates a continuous tone.
 - **off_time** - the duration when no tone is played.

The following are examples of DIAL_TONE:

1. 350,440;-1 (350 and 440 Hz continuous tone)

- **RINGING_TONE [frequency1 | frequency2 | on_time | off_time]** is used to select the tone advising the caller that a connection is made and a calling signal is applied to a telephone number or service point. You can select the country specific tone. The default tone is the North American tone.
 - **frequency1** - the frequency of tone 1.
 - **frequency2** - the frequency of tone 2.
 - **on_time** - the duration of the tone when it is on. A -1 indicates a continuous tone.
 - **off_time** - the duration when no tone is played.

The following are examples of RINGING_TONE:

1. 440,480; 2000,4000 (440 and 480 Hz with 2 seconds on, 4 seconds off)

- **BUSY_TONE [frequency1 | frequency2 | on_time | off_time]** is used to select the tone advising the caller that the telephone number is busy. You can select the country specific tone. The default tone is the North American tone.
 - **frequency1** - the frequency of tone 1.
 - **frequency2** - the frequency of tone 2.
 - **on_time** - the duration of the tone when it is on. A -1 indicates a continuous tone.
 - **off_time** - the duration when no tone is played.
- **FASTBUSY_TONE [frequency1 | frequency2 | on_time | off_time]** is used to select the tone advising the caller that the telephone number is busy. It is fast in cadence or frequency. You can select the country specific tone. The default tone is the North American tone.
 - **frequency1** - the frequency of tone 1.
 - **frequency2** - the frequency of tone 2.
 - **on_time** - the duration of the tone when it is on. A -1 indicates a continuous tone.
 - **off_time** - the duration when no tone is played.
- **CONGESTION_TONE [frequency1 | frequency2 | on_time | off_time]** is used to select the tone advising the caller that the groups of lines or switching equipment necessary for setting up the required call, or for the use of a specific service, are temporarily engaged. You can select the country specific tone. The default tone is the North American tone.
 - **frequency1** - the frequency of tone 1.
 - **frequency2** - the frequency of tone 2.
 - **on_time** - the duration of the tone when it is on. A -1 indicates a continuous tone.
 - **off_time** - the duration when no tone is played.

The IP Phone 1140E supports using WAV files to replace the ringtone Frequency/Cadence pattern. For a system-wide setting, the country default values can be used.

NAT configuration commands

- **NAT_SIGNALLING [NONE | SIP_PING | STUN]** indicates the type of protocol used for NAT traversal in the signaling port. The IP Phone 1140E with SIP Firmware supports two methods of NAT traversal of the signaling path: SIP_PING and STUN.

- **NONE** - If the value is not set to None, this parameter overrides the value of the parameter SIP_PING in the device configuration file.
- **SIP_PING** - a Nortel proprietary NAT traversal protocol. Note that SIP_PING only supports NAT traversal in the signaling port.
- **STUN** - the most common NAT traversal method.
- **NAT_MEDIA [NONE | STUN]** indicates the type of protocol used for NAT traversal in the media ports. The default is NONE.
 - **NONE** - is the default and disables NAT_MEDIA.
 - **STUN** - the most common NAT traversal protocol for the media (RTP and Real-time Control Protocol [RTCP]) port.
 - **x** - is the binding lifetime in seconds.

ATTENTION

NAT_TTL [x] is used for future development. Currently, the default value is 2 minutes (120 seconds) and SIP IP Phones 1120E/1140E do not process or use the value defined in **NAT_TTL [x]**. SIP IP Phones 1120E/1140E always ping the ports at regular intervals of 60 seconds regardless of the NAT_TTL value.

- **STUN_SERVER_IP1[ip_address]** NAT traversal using STUN protocol requires a STUN server in the public internet. Two STUN server IPs can be provisioned.
 - **ip_address** - is the IP address of STUN server 1.
- **STUN_SERVER IP2[ip_address]** NAT traversal using STUN protocol requires a STUN server in the public internet. Two STUN servers IPs can be provisioned.
 - **ip_address** - is the IP address of STUN server 2.
- **STUN_SERVER_PORT1[port_number]** is the port number used corresponding to STUN_SERVER_IP1. The default port number is 3478.
 - **port_number** - is the port number.
- **STUN_SERVER_PORT2[port_number]** is the port number used corresponding to STUN_SERVER_IP2. The default port number is 3478.
 - **port_number** - is the port number.

VQMon configuration commands

It is important to read "[How VQMon works](#)" (page 92) before configuring the VQMON parameters.

- **VQMON_PUBLISH [YES | NO]** is the command that is used to enable or disable the publish message containing the voice quality monitoring metrics sent to the Proactive Voice Quality Monitoring (PVQMoN) collecting server.
 - **YES** - enables VQMoN.
 - **NO** - disables VQMoN. NO is the default.

- **VQMON_PUBLISH_IP [xxx.xxx.xxx.xxx]** is used to set the IP address of the PVQMoN server that collects voice quality monitoring metrics from the publish message.

This IP address is used within the report only.

- **LISTENING_R_ENABLE [YES | NO]** is used to enable or disable the alerts based on the Listening R Minor and Major Thresholds. The default value is vocoder-dependent using a scale from 1 (lowest quality) to 100 (highest quality). Currently default values are used based on VOCODER on a per call basis as summarized below.
 - **YES** - enables the sending of the alert report based on the Listening R Value.
 - **NO** - disables the sending of the alert report based on the Listening R Value.

VOCODER_G711_ULAW VOCODER_G711_ULAWPLP	LISTENING_R_WARN = 80 LISTENING_R_EXCE = 70
VOCODER_G723 VOCODER_FLAG_G723_RATE_53 VOCODER_FLAG_G723_RATE_63	LISTENING_R_WARN = 60 LISTENING_R_EXCE = 50
VOCODER_G729 VOCODER_G722 VOCODER_PCM16 VOCODER_PCM8 vqmonVocoderTypeUnknown	LISTENING_R_WARN = 70 (default if not configured and unknown type) LISTENING_R_EXCE = 60

- **LISTENING_R_WARN [xx]** is the threshold to send a report on Listening R less than [xx]. The default value is 70. Using 0 will reset it to default based on far end VOCODER.
 - **xx** - is an INTEGER value used as threshold.

- **LISTENING_R_EXCE [xx]** is the threshold to send a report on Listening R less than [xx]. The default value is 60. Using 0 will reset it to default based on far end VOCODER.
 - **xx** - is an INTEGER value used as threshold.
- **PACKET_LOSS_ENABLE [YES | NO]** is used to enable or disable the alerts based on the packet loss thresholds. Packet loss is the fraction of RTP data packets from the source lost since the beginning of reception. The value is an integer scaled by 256. The range is 1 to 25600.
 - **YES** - enables the sending of alert report based on the packet loss
 - **NO** - disables the sending of alert report based on the packet loss
- **PACKET_LOSS_WARN [xx]** is the threshold to send a report on Packet Loss greater than [xx]. The default is 256 (1%). Using 0 will reset the threshold to default.
 - **xx** - is an INTEGER value scaled by 256 that is used as threshold. The range is 1 to 25600.
- **PACKET_LOSS_EXCE [xx]** is the threshold to send a report on Packet Loss greater than [xx]. The default is 1280 (5%). Using 0 will reset the threshold to default.
 - **xx** - is an INTEGER value scaled by 256 that is used as threshold. The range is 1 to 25600.
- **JITTER_ENABLE [YES | NO]** is used to enable or disable alerts based on the inter-arrival Jitter on incoming RTP packets inter-arrival time. The value is represented in 1/65536 of a second.
 - **YES** - enables the sending of alert report based on jitter detection
 - **NO** - disables the sending of alert report based on jitter detection
- **JITTER_WARN [xx]** is the threshold to send a report on Inter-arrival Jitter greater than [xx]. 1 second is broken up into 65535 (0xffff hex) parts. [xx] / 65535 is the threshold in seconds. The default is 3276 (50 ms). Using 0 will reset the threshold to default.
 - **xx** - is an INTEGER value used as threshold
- **JITTER_EXCE [xx]** is the threshold to send a report on Inter-arrival Jitter greater than [xx]. 1 second is broken up into 65535 (0xffff hex) parts. [xx] / 65535 is the threshold in seconds. The default is 32760 (500 ms). Using 0 will reset the threshold to default.
 - **xx** - is an INTEGER value used as threshold

- **DELAY_ENABLE [YES | NO]** is used to enable or disable the alerts based on the excessive delay detection. This is the one-way delay (including system delay) for the call, measured in milliseconds.
 - **YES** - enables Excessive delay detection.
 - **NO** - disables Excessive delay detection.
- **DELAY_WARN [xx]** is the threshold to give warning on Excessive Delay greater than [xx]. The default is 150 ms. Using 0 will reset the threshold to default.
 - **xx** - is an INTEGER value used as a threshold measured in 1/1000 of a second.
- **DELAY_EXCE [xx]** is the threshold to report unacceptable Excessive Delay greater than [xx]. The default is 175 ms. Using 0 will reset the threshold to default.
 - **xx** - is an INTEGER value used as a threshold measured in 1/1000 of a second.
- **SESSION_RPT_EN [YES | NO]** is used to enable or disable periodic VQMon session reports. The default is disabled.

Both session report enable and session report interval must be set if the phone firmware has been upgraded to SIP Release 2.0. Otherwise, the SESSION_RPT_INT default of 60 seconds will be used automatically.

 - **YES** - enables periodic VQMon session reports.
 - **NO** - disables periodic VQMon session reports. Default is NO.
- **SESSION_RPT_INT [xx]** is used to specify the interval for the periodic VQMon session report in seconds. The minimum acceptable value is 60 seconds. The maximum acceptable value is 600 seconds. The default is 60 seconds.
 - **xx** - is an INTEGER value in seconds.

System commands

- **ADMIN_PASSWORD [password]** is used to change the default administrator password of the IP Phone 1140E that is used for unlocking network menus. The default is 26567*738.
 - **password** - the administrator password.

Create the set configuration file on the provisioning server

If the IP Phone 1140E encounters a [USER_CONFIG] section while parsing the 1140eSIP.cfg configuration file, the set will download its set specific configuration file SIP<mac id>.cfg.

Set specific configuration files support customizing the set on a per set/user level. Parameters in the device configuration file can be overwritten with a set specific configuration file.

Most of the parameters in the set configuration file are saved on the set. Removing a parameter from the set configuration file does not change the parameters saved on a configured set. If a parameter is set only in the set specific configuration file, removing the set specific configuration file will not clear the setting.

Note: If the 1140eSIP.cfg configuration file contains a [USER_CONFIG] section, it is recommended that DOWNLOAD_MODE be set to FORCED. This is a global setting for all sets used to determine if the mac id file should be read. Alternatively, if the user wishes to use DOWNLOAD_MODE set to AUTO, when a change is made to any mac id file the version number should be incremented so that all phones read the file.

Table 7 "Set configuration commands" (page 60) provides a summary of the commands that can be used in the set configuration file. The syntax of each command is summarized in "Set configuration commands" (page 60).

Table 7
Set configuration commands

Auto login	AUTOLOGIN_ID_KEY01 AUTOLOGIN_PASSWD_KEY01
------------	--

Set configuration commands

- **AUTOLOGIN_ID_KEY01** [* | xx] This parameter is located within the set specific configuration file. This is the id the set will use to register and authenticate. The default user id "user1" will be used, if an id is not supplied and the set has not previously logged in.
 - * - indicates that the set should use its mac address (lower case) as the user id
 - xx - an ASCII string that corresponds to the user id.

Note: To provision AUTOLOGIN_ID_KEY01, the IP Phone 1140 must be rebooted after the set configuration file is updated. To force a hard reboot after the set configuration file is updated set FORCE_REBOOT YES in the device configuration file.

- **AUTOLOGIN_PASSWD_KEY01** This parameter is located within the set specific configuration file. There is not a default password. If this is

blank and AUTOLOGIN_ENABLE is set to USE_AUTOLOGIN_ID (or 2) in the device configuration file, the set will not login.

Note: To provision AUTOLOGIN_PASSWD_KEY01, the IP Phone 1140 must be rebooted after the set configuration file is updated. To force a hard reboot after the set configuration file is updated set FORCE_REBOOT YES in the device configuration file.

Create the Dialing Plan file on the provisioning server

A dialing plan essentially describes the number and pattern of digits that a user dials to reach a particular telephone number. Access codes, area codes, specialized codes, and combinations of the number of digits dialed are all part of a dialing plan.

The purpose of the dialing plan is so that the end user does not have to press the send or pound key (#) to have the IP Phone 1140E with SIP Firmware send the initial message to start the call.

Dialing a telephone number on an IP Phone that supports SIP can be different than dialing a number from a traditional telephone. SIP signaling is communicated through a SIP URI to get to the far end. For example, you can key in the SIP address, jsmith@yourcompany.com to reach John Smith. When the IP Phone 1140E with SIP Firmware receives this address, the dialing plan is bypassed and the IP Phone 1140E simply uses the SIP URI to send a SIP INVITE to jsmith@yourcompany.com (INVITE sip:jsmith@yourcompany.com).

Entering a SIP URI address, however, is inconvenient for an IP Phone with SIP Firmware unless a USB keyboard is attached. Also, the user must explicitly press the send key (or use some method to indicate the end of the URI) to indicate the completion of the SIP address. This is not something that the user is accustomed to in a traditional PBX environment.

The alternative is to use a URI where numbers are used to reach the far end. Using different access codes, the IP Phone with SIP Firmware translates the digits entered into something that the server can understand and remaps the number entered into different URIs. Some of the numbers are mapped as intercom calls, some numbers are mapped as local Public Switched Telephone Network (PSTN) calls, and some numbers are mapped as public long-distant calls.

The issue is that until the IP Phone itself can determine the type of call, no SIP INVITE message is sent. This is where the dialing plan comes into effect. The call type is determined by the dialing plan. Based on the rules defined in the dialing plan, once a match has been identified, the IP Phone 1140E with SIP Firmware sends the invite without the need to press the send key. This behavior closely matches the traditional PBX operation.

The IP Phone 1140E with SIP Firmware design places no restriction in the format of the SIP URI. The dialing plan is a scheme to match the user experience with traditional PBX operation. It does not restrict the type of URI that the user can use.

The IP Phone 1140E with SIP Firmware uses a dialing plan to recognize a call as an emergency call when it sends an INVITE. The dialing plan can have multiple emergency numbers. See the chapter ["Emergency 911 - Operator control of disconnect"](#) (page 125) for information on the handling of Emergency 911 calls by the IP Phone 1140E with SIP firmware.

The following is an example of a dialing plan.

Figure 6
Sample dialing plan

```

/* ----- */
/* A simple dial plan */
/* ----- */
$N="yourcompany.com"
$t=300
$s=0

%%

/* DIGITMAP: Operator call */

(0)(0)# && sip:$N;$N;user=phone &&

/* DIGITMAP: Help Desk */

(411)(411)# && sip:$N;$N;user=phone &&

/* DIGITMAP: Emergency call */

(911)(911)# && sip:$N;$N;user=phone && t=100

/* DIGITMAP: Private intra-location call, no access code */

([0496]x{3})([0496]x{3})# && sip:$N;$N;user=phone &&

/* DIGITMAP: Public local call, access code 9 */

(9[1]x{9})(9[1]x{9})# && sip:$N;$N;user=phone &&

/* DIGITMAP: Private Intra-company Call, access code 6 */

(6[10]x{6})(6[10]x{6})# && sip:$N;$N;user=phone &&

/* DIGITMAP: Public national call, access code 61 */

(61x{10})(61x{10})# && sip:$N;$N;user=phone &&

/* DIGITMAP: Public international call, access code 6011 */

(6011x{7,15})(6011x{7,15})# && sip:$N;$N;user=phone && t=8000

```

Dialing function description

Dialing plan

As most telephone users are used to dialing digits to indicate the address of the destination, there is a need to specify the rule by which digits are transformed into a URI. The IP Phone 1140E with SIP Firmware dialing plan contains two sections delimited by two percent signs (%%).

Figure 7
Sample dialing plan declarations section

```
+-----+
| declarations section | user pre define variables and parameters |
| %%                  | section separator |
| digit maps          | list of digit maps |
+-----+
```

In the declaration section, the administrator can define the variables. The variables must start with a dollar (\$) sign, followed by a number or a character, such as \$1 or \$a. There are two variables that are reserved by system. They are as follows:

\$: used for the collected digits if they match the pattern

\$t : default timer

There must be a domain name defined and the domain name can be represented by any variable. In [Figure 6 "Sample dialing plan" \(page 62\)](#), the domain name is represented by \$n.

The variable definitions take the form:

Figure 8
Sample dialing plan variable definitions

```
+-----+
| Name = value |
+-----+
```

For example:

\$1="nortel.com"

\$2="Nortel"

```

$3="."
$4="com"
$5="Nortel.com"
$t=10000 (default timer is 10 seconds)
$a=Nortel.com

```

The second section of dialing plan contains the digit map. The digit map section has three subsections that are divided by a separator of two ampersands (&&).

Figure 9
Sample dialing plan digit map section

```

+-----+
| patterns && destination string && dialing action attributes |
+-----+

```

The first part of a dialing plan contains a pattern defined with DRegex, which is used for matching the dialed number. The patterns are separated by the pipe (|) sign. The second part contains the result string used in the dial step. The third part defines the parameters used by UA in dialing action.

The following parameter is currently defined:

`t=xxxx`: After this timer expires, the number entered is automatically dialed. The timer starts after the first digit is entered and after it expires, the collected digits are automatically dialed out. `xxxx` is a decimal number in msec. The default timer is used when `t` is not specified in the digit map.

For example:

```
X{4} && sip:$$. phone-context=nortel.com;user=phone && t=7000
```

When the user presses any 4 digits, such as 4567, the following SIP URIs are generated because of the translation rule:

`Sip:4567; phone-context=nortel.com;user=phone`. The timeout of stopping the collection of digits is 7 seconds.

The pound sign (#) at the end of the digit map causes the IP Phone 1140E to dial the matched dialing plan immediately.

DRegex

The Digit Regular Expression (DRegex) syntax is a telephony-oriented mapping of Portable Operating System Interface (POSIX) Extended Regular Expressions (ERE). Users must take care not to confuse the DRegex syntax with POSIX EREs as they are not identical. In particular, there are many features of POSIX EREs that DRegex does not support. The dialing plan uses DRegex instead of ERE. The following rules demonstrate the use of DRegex.

Figure 10
DRegex rules

Entity	Matches
character	digits 0-9, *, #, and A-D (case insensitive, A-D only for military requirements)
*	the * character
#	the # character
[character selector]	Any character in selector
[^digit selector]	Any digit (0-9) not in selector
[range1-range2]	Any character in range from range1 to range2, inclusive
x	Any digit 0-9
{m}	m repetitions of previous pattern
{m,}	m or more repetitions of previous pattern
{,n}	At most n (including zero) repetitions of previous pattern
{m,n}	at least m and at most n repetitions of previous pattern
()	provide "captures" for back reference variable \$\$
\$\$	back reference "matches" text previously matches within parentheses or the "matches" if parentheses is not specified
/* comments line */	comments

Example	Description
1	Matches the digit 1
[179]	Matches 1, 7, or 9
[2-9]	Matches 2, 3, 4, 5, 6, 7, 8, 9
[^15]	Matches 0, 2, 3, 4, 6, 7, 8, 9
[02-46-9A-D]	Matches 0, 2, 3, 4, 6, 7, 8, 9, A, B, C, D
x	Matches 0, 1, 2, 3, 4, 5, 6, 7, 8, 9
*6[179#]	Matches *61, *67, *69, or *6#
x{10}	Ten digits (0-9)
011x{7,15}	011 followed by seven to fifteen digits
91(x{10})	matches 91 followed by 10 digits (does not include 91)
	Ex: 911234567890
	\$\$=1234567890

Downloadable WAV files

It is possible to customize the ring tones on the IP Phone 1140E with SIP Firmware. Up to five special ring tones can be downloaded from the provisioning server and stored on the IP Phone 1140E. The end user can select which ring tone they would like to implement.

In order to download these special files, the files must reside on the provisioning server and be specified in the SIP provisioning file. For more information, see ["Download the SIP Firmware to the provisioning server" \(page 30\)](#). The WAV files have a maximum size of 512 KB each for the IP Phone 1140E.

The file format is restricted to ITU-T A-law or u-law (8.0 kHz, 8-bit, mono or 16.0 kHz, 16 bit mono).

After the WAV files are downloaded to the IP Phone 1140E, the WAV file names appear in **Pref, Audio, Tones, Ring Pattern** (1 to 8 are standard ring tones, and 9 and above are WAV ring tones) and the WAV ring tones can then be selected to replace the standard ring tones.

For further information about downloadable WAV files, see the *SIP Firmware Release 2.0 for IP Phone 1140E User Guide* (NN43113-101).

Configure the DHCP server

Use DHCP to provide partial Device Settings or full Device Settings, as well as to automatically provide Voice VLAN IDs.

Table 8
Characteristics of Partial DHCP mode and Full DHCP mode

Partial DHCP mode	Partial DHCP mode retrieves the following network parameters from the DHCP server: IP address, subnet mask, and default gateway configuration for the IP Phone 1140E.
Full DHCP mode	As well as those parameters retrieved in Partial DHCP mode, Full DHCP also retrieves the following from the DHCP server: DNS server and Provisioning Server.
Auto VLAN Discovery	Auto VLAN Discovery retrieves the Voice VLAN ID for the IP Phone 1140E with SIP Firmware.

Partial DHCP mode

After the IP Phone 1140E with SIP Firmware is configured to operate in Partial DHCP mode, the DHCP server does not need any special configuration to support an IP Phone 1140E with SIP Firmware. The IP Phone 1140E with SIP Firmware receives the following Device Settings parameters from the DHCP server in the following table:

Table 9
DHCP options in Partial DHCP mode

Parameter requested by the IP Phone 1140E	Description	DHCP server option
Subnet mask	The subnet mask of the IP Phone 1140E.	1
Router and gateways	IP address of the default gateway of the IP Phone 1140E.	3

Lease time	Implementation varies according to DHCP server configuration.	51
Renewal time	Implementation varies according to DHCP server configuration.	58
Rebinding interval	Implementation varies according to DHCP server configuration.	59

Partial DHCP is the response from a DHCP server that is not configured to recognize the Nortel-specific Vendor Class Identifier in option 60 (Nortel-SIP-Phone-A). After the IP Phone 1140E is configured to operate in Partial DHCP mode, the DHCP server needs no special configuration to support an IP Phone 1140E. Using Partial DHCP, an IP Phone 1140E can obtain its IP address, subnet mask, and gateway IP address. The remainder of the configuration information is manually entered on the IP Phone 1140E. This includes the DNS server IP and the Provisioning server address and protocol.

Full DHCP mode

The DHCP server requires special configuration in Full DHCP mode. The IP Phone 1140E with SIP Firmware obtains Device Settings parameters from specially configured DHCP servers.

The IP Phone 1140E with SIP Firmware requests the following Device Settings parameters from the DHCP server:

- IP address configuration for the IP Phone 1140E
- subnet mask for the IP Phone 1140E IP address
- default gateway for the IP Phone 1140E subnet
- DNS server
- Provisioning Server

It is also possible to obtain the DNS IP automatically using Full DHCP mode without any special configuration of the DHCP server. This means that there is no need to configure the Nortel-specific Vendor Class Identifier (Option 60) on the DHCP server. However, the provisioning server address needs to be manually defined within the Device Settings menu. This configuration requires toggling to Partial DHCP mode to define the provisioning server address and toggling back to Full DHCP mode after this is complete.

Configure the DHCP server to support Full DHCP mode

SIP IP Phone class identifier

A Nortel-SIP-Phone-aware DHCP server is needed for Full DHCP mode. For Full DHCP mode, the DHCP server requires special configuration.

After the DHCP server is configured to recognize the IP Phone 1140E with SIP Firmware as a unique IP Phone, the DHCP server can treat the IP Phone 1140E differently than other DHCP IP Phones. An IP Phone-aware DHCP server can automatically configure Nortel IP Phones by sending all information that the IP Phone requires.

The IP Phone 1140E and the DHCP server communicate using a unique class identifier. After the IP Phone 1140E first sends the DHCP DISCOVER, it includes the Nortel-SIP-Phone-A ASCII string within the Vendor Class Identifier (Option 60). The DHCP server recognizes this special Vendor Class Identifier (Option 60) and sends back OFFER, which also includes the same Vendor Class Identifier. This makes it possible to notify the IP Phone 1140E with SIP Firmware that the server is IP Phone-aware, and that it is safe to accept the offer from the server.

Every Nortel IP Phone 1140E with SIP Firmware fills in the Vendor Class ID option of the DHCPDISCOVER and DHCPREQUEST messages with the null-terminated, ASCII-encoded string Nortel-SIP-Phone-A, where A identifies the version number of the information format of the IP Phone.

The Class Identifier Nortel-SIP-Phone-A must be unique in the DHCP server domain.

The unique DHCP configuration is required to allow the DHCP server to respond with a unique Option 66 parameter to the IP Phone 1140E with SIP Firmware.

Note: The DHCP standard defines Option 66 as the bootp server address in a string. The meaning of the bootp server address is extended in Nortel IP Phone 1140E with SIP Firmware to include the provisioning server address. The string in the DHCP offer for Option 66 can be

- the numeric IP address or name of the TFTP server or the URI (if FTP or HTTP protocol is used) of the provisioning server in the form of

<protocol>://<provisioning server URL>

For example:

http://mydomain.com/SIP_phone

If provisioning server authentication is required, the user credential must be embedded in the URI in the form of

<protocol>://[<userid>:<password>@]<provisioning server URL>[:port][path]

For example,

ftp://www.mydomain.com/ABC or ftp://myuserid:mypass@ftp.mydomain.com:21/ABC

Requested Device Settings parameters

Using Full DHCP mode, a SIP IP Phone-aware DHCP server can automatically configure Nortel SIP IP Phones by requesting a list of configuration parameters. The IP Phone 1140E uses DHCP to request and receive the information.

The following table lists the Device Settings parameters requested by the IP Phone 1140E in the Parameter Request List option (Option Code 55) in the DHCPDISCOVER and DHCPREQUEST messages. The DHCPOFFER and the DHCPACK reply messages from the DHCP server must contain the options in the following table.

Table 10
DHCP options in Full DHCP mode

Parameter requested by the IP Phone 1140E	Description	DHCP server option
Subnet mask	This is the subnet mask of the IP Phone 1140E	1
Router and gateways	IP address of the default gateway of the IP Phone 1140E.	3
DNS server	DNS Server address; only the first one from the list is used.	6
Broadcast address	This is the Broadcast address of the subnet. The IP Phone 1140E automatically calculates the Broadcast address if it is not provided.	28
DNS domain	Implementation varies according to DHCP server.	15
Lease time	Implementation varies according to DHCP server.	51
Renewal time	Implementation varies according to DHCP server.	58

Rebinding interval	Implementation varies according to DHCP server.	59
Provisioning server	Used for delivering the provisioning server IP address. This parameter can contain either IP address or a URL to the provisioning server or folder.	66

DHCP VLAN Auto Discovery

Configuring a server for Voice VLAN Discovery is optional. This configuration is done in addition to any configuration done for Full DHCP or Partial DHCP and it is required only when you are configuring the VLAN Auto Discovery in the Device Settings menu on the IP Phone 1140E.

Auto Voice VLAN Discovery is only possible in Full and Partial DHCP mode.

VLAN Auto Discovery configuration is useful in a network with separate Data VLAN for traffic (commonly used for PC-to-PC communication) and Voice VLAN for VoIP traffic with different priorities.

The VLAN Auto Discovery is a two-step process:

1. DISCOVER is sent to the DHCP server for available Voice VLAN IDs. The DHCP server sends an OFFER with the available Voice VLAN IDs. If the Data VLAN ID has been manually provisioned in the Device Settings of the IP Phone 1140E, DHCP DISCOVER is tagged with the Data VLAN ID; otherwise, it is untagged.
2. DISCOVER is sent to the DHCP server for all of the DHCP required parameters. However, this DISCOVER is tagged with the Voice VLAN obtained in step 1.

DHCP VLAN Auto Discovery requires a Nortel-SIP-Phone-aware DHCP server. All DHCP requests carry the Vendor Class Identifier, Nortel-SIP-Phone-A, to allow the DHCP server to identify that the requests are coming from an IP Phone 1140E with SIP Firmware.

DHCP Auto Discovery returns Voice VLAN IDs. The DHCP protocol provides no standard option for VLAN ID requests. Separate DHCP vendor-specific entry is needed for DHCP VLAN Auto Discovery to convey the VLAN information to the IP Phone. DHCP VLAN Auto Discovery uses one of the reserved for site-specific use DHCP options for VLAN list retrieval. At least one of the following Nortel site-specific options must be returned by the DHCP server as part of each DHCPOFFER and DHCPACK message for the IP Phone to accept these messages as valid; 43, 128, 131, 144, 157, 188, 191, 205, 219, 223, 232, 247, 251.

After multiple VLAN IDs are returned from the DHCP server, the IP Phone 1140E tries to connect to each of the VLANs, following the order in which VLAN IDs are specified in the DHCP option.

The format of the field for DHCP VLAN Auto Discovery is: Type, length, and data, described in the following sections.

Type (1 octet)

To avoid the possibility of option types already being used by different vendors, there are fourteen options types supported by the telephone. They are: 128, 131, 144, 157, 188, 191, 205, 219, 223, 232, 247, 251, 247, and 251. Select one value from the type byte list for the DHCP OFFER response. Nortel recommends using 232, 247, or 251. DHCP option numbers less than 224 are reclaimed by the IETF (RFC 3942). Future changes in the DHCP protocol can force the telephone to stop sending these option requests. Currently, IP Phone 1140E with SIP Firmware does support the remaining listed options to maintain backward compatibility.

Length (1 octet)

The Length value is variable. Count only the number of octets in the data field.

Data (variable number of octets)

ASCII based format: VLAN-A:XXX+YYY+ZZZ. where, VLAN– A: uniquely identifies this as the Nortel DHCP VLAN discovery request. Each VLAN ID is followed by a plus (+) sign if there are more VLAN IDs or a period (.) to terminate the string.

There are a maximum of 10 VLAN IDs that can be configured in the current version.

Once IP Phone 1140E with SIP Firmware receives the DHCP offer containing the site-specific Voice VLAN option, the next DHCPDISCOVERY message is tagged with the Voice VLAN ID.

Physically install the IP Phone 1140E

Complete instructions to physically install the IP Phone 1140E, including detailed figures and applicable warnings, are given in the document *SIP Firmware Release 2.0 for IP Phone 1140E User Guide* (NN43113-101).

The steps for installing the IP Phone 1140E are summarized in the following procedure.

Procedure 4

Installing the IP Phone 1140E

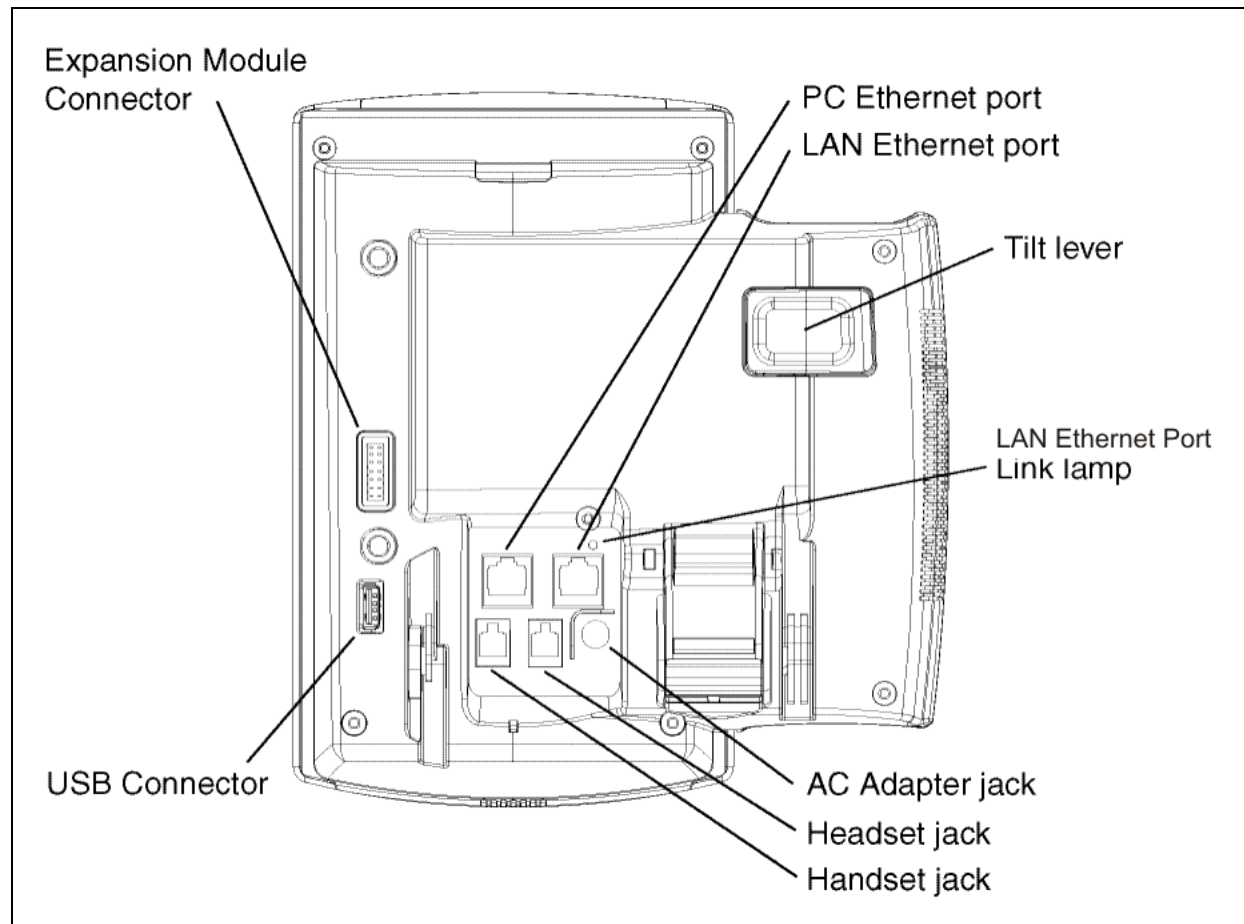
Step	Action
1	Remove the stand cover. Pull upward on the center catch and remove the stand cover. The cable routing tracks are now accessible.
2	Connect the AC power adapter (optional). Connect the adapter to the AC adapter jack in the bottom of the IP Phone 1140E. Form a small bend in the cable, and then thread the adapter cord through the channels in the stand.
3	Install the handset. Connect the end of the handset cable with the short straight section into the handset. Connect the end of the handset cable with the long straight section to the back of the IP Phone 1140E, using the RJ-9 handset jack. Form a small bend in the cable, and then thread the handset cord through the channels in the stand so that it exits behind the handset on the right side, in the handset cord exit in the stand base.
4	Install the headset (optional). If installing a headset, plug the connector into the RJ-9 headset jack on the back of the IP Phone 1140E, and thread the headset cord along with the handset cord through the channels in the stand, so that the headset cord exits the channel.
5	Install the Ethernet cable. Connect one end of the supplied Ethernet cable to the back of the IP Phone 1140E using the RJ-45 connector and thread the network cable through the channel.

- 6 Install the Ethernet cable connecting the PC to the IP Phone 1140E (optional). If connecting PC Ethernet through the IP Phone 1140E, connect one end of the PC Ethernet cable to the IP Phone 1140E using the RJ-45 connector and thread it through the channel. Connect the other end to the LAN connector on the back of the PC.
- 7 Install additional cables. If applicable, plug in optional USB devices. Connect the Ethernet cable to the LAN Ethernet connection. If using an AC power adapter, plug the adapter into an AC outlet.
- 8 Wall-mount the IP Phone 1140E (optional). The IP Phone 1140E can be mounted either by: (method A) using the mounting holes on the bottom of the IP Phone 1140E stand, or (method B) using a traditional-style wall-mount box with RJ-45 connector and 15-cm (6-inch) RJ-45 cord (not provided).
- 9 Replace the stand cover. Ensure that all cables are neatly routed and press the stand cover into place until a click is heard.
- 10 Put the IP Phone 1140E in the wall-mount position (optional). If the IP Phone 1140E is to be mounted on the wall, put it in the wall-mount position by holding the tilt lever and pressing the IP Phone 1140E towards the base until the IP Phone 1140E is parallel with the base. Release the tilt lever and continue to push the IP Phone 1140E towards the base until an audible click is heard. Ensure the IP Phone 1140E is securely locked in position.

—End—

The following figure shows the connections on the IP Phone 1140E.

Figure 11
IP Phone 1140E connections



Upgrade and convert the IP Phone 1140E firmware

Introduction

This chapter describes how to upgrade an IP Phone 1140E with UNISim firmware to SIP firmware.

In order to upgrade an IP Phone 1140E with UNISim firmware, first determine if you have the minimum UNISim firmware release on the IP Phone (0625C39). If your IP Phone 1140E is installed with the minimum version of UNISim firmware, proceed to the section "[Convert UNISim Firmware to SIP Firmware on the IP Phone 1140E](#)" (page 87). If your IP Phone 1140E is not installed with the minimum version of UNISim firmware, proceed to the section "[Upgrade to the minimum UNISim Firmware](#)" (page 81).

To convert the firmware on the IP Phone 1140E from SIP to UNISim, see the section "[Maintenance and troubleshooting](#)" (page 149).

Upgrade the SIP Firmware on the IP Phone 1140E

Use the following procedures to upgrade existing SIP Firmware to new SIP Firmware on the IP Phone 1140E.

Download the SIP Firmware to the provisioning server

To download the SIP Firmware, follow the steps in the next procedure.

Procedure 5

Downloading SIP Firmware for the IP Phone 1140E from the Nortel Web site

Step	Action
1	Go to www.nortel.com/support .
2	Log on to the Nortel Web site with a valid Nortel User ID and Password. The Technical Support page appears.

- 3 Enter IP Phone 1140E in the **Knowledge and Solution Engine** box.
- 4 Select **Software** in the **All types** scroll down menu.
- 5 Press the gray arrow at the end of the **Knowledge and Solution Engine** box to obtain the **Search Results**.
- 6 From the **Search Results**, select the appropriate version of the SIP Firmware for the IP Phone 1140E, for example, **SIP IP Phone 1140E Release 0625C39D26.bin**.
- 7 Place the selected firmware on the provisioning server.

—End—

Modify the SIP provisioning file

Use the following procedure to modify the SIP provisioning file, which exists on the provisioning server.

Procedure 6

Modifying the SIP provisioning file

Step	Action
1	Under the firmware [FW] section of the SIP Provisioning file, increase the VERSION number (for example 06A5C39d26).
2	Under the firmware [FW] section of the SIP Provisioning file, modify the FILENAME of the new file you want to upload to the IP Phone 1140E.
3	<p>Invoke the upgrade mechanism.</p> <p>Use one of the next three methods to invoke a firmware upgrade on the IP Phone 1140E with SIP Firmware.</p> <ol style="list-style-type: none"> 1. Power off and power on the IP Phone 1140E. 2. Select Services, System, Check For Updates on the IP Phone 1140E. 3. Allow for an automatic check for updates to occur. (See AUTO_UPDATE under "Feature configuration commands" (page 42)).

Any of these actions causes the IP Phone 1140E to contact the provisioning server and attempt to read the Provisioning file. A Soft Reset (**Srvcs, System, Reset Phone**) does not cause the IP Phone 1140E to retrieve the Provisioning file and hence does not cause a firmware upgrade.

—End—

Upgrade to the minimum UNISTim Firmware

The IP Phone 1140E can be ordered with UNISTim Firmware installed or with SIP Firmware installed. You can convert the firmware on an IP Phone 1140E from UNISTim to SIP. To successfully convert the firmware from UNISTim to SIP, the UNISTim Firmware version on your IP Phone 1140E must be 0625C39 or higher.

Identify the current version of UNISTim Firmware

For a new IP Phone 1140E, follow [Procedure 7 "Checking the UNISTim Firmware version on a new IP Phone 1140E" \(page 81\)](#) to determine the version number of the UNISTim Firmware on an IP Phone 1140E.

For an IP Phone 1140E already in use, follow [Procedure 8 "Checking the UNISTim Firmware version on an IP Phone 1140E already in use" \(page 82\)](#) to determine the version number of UNISTim Firmware on an IP Phone 1140E.

Procedure 7

Checking the UNISTim Firmware version on a new IP Phone 1140E

Step	Action
1	<p>After assembling the IP Phone 1140E and turning it on, the display on the IP Phone 1140E goes through the following sequence:</p> <ul style="list-style-type: none"> • Nortel splash screen appears • Nortel sonic sound plays • Nortel banner appears <p>Following the Nortel banner, the firmware version appears in the display (F/W version).</p>
2	<p>Note the UNISTim Firmware version number and write it down. Compare the version number to the minimum-required UNISTim Firmware version (0625C39).</p> <p>UNISTim Firmware version names contain numbers and letters. Use the last three characters in a version to compare the version of UNISTim on an IP Phone 1140E (0625C39) with the minimum required version for the upgrade. Note that C23 is greater than C39 and C1B is less than C39.</p>

If the version number is equal to or higher than 0625C39, go to the section "[Convert UNISTim Firmware to SIP Firmware on the IP Phone 1140E](#)" (page 87).

If the number is lower than 0625C39, go to the section "[Upgrade UNISTim Firmware to the minimum required UNISTim Firmware](#)" (page 84) and follow the instructions to upgrade an IP Phone 1140E to the minimum-required version of UNISTim Firmware before a conversion to SIP Firmware.

—End—

Procedure 8

Checking the UNISTim Firmware version on an IP Phone 1140E already in use

Step	Action
------	--------

- | | |
|---|--|
| 1 | Press the Globe/Services key on the IP Phone 1140E twice quickly. |
|---|--|



If the admin password prompt appears, enter the password **26567*738**.

The Local Tools menu appears:

Table 11
Local Tools menu

- | |
|--------------------------|
| 1. Preferences |
| 2. Local Diagnostics |
| 3. Network Configuration |
| 4. Lock Menu |

To make a selection, press the number associated with the menu item, or use the **Navigation key cluster**.

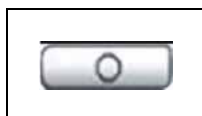


to scroll through the menu items. Press the **Select** key to select the highlighted menu item.

Table 12
Using the Navigation key cluster to navigate in the Local Tools menu

Key	Action
Down	Moves highlight down
Up	Moves highlight up
Right	Selected current menu item
Left	Closes menu
Select key (center of cluster)	Selects current menu item

To close this menu, use the **Quit** key.



- 2 Select **2. Local Diagnostics** in the Local Tools menu by pressing the **Select** key in the Navigation key cluster or by pressing the number **2**.
- 3 Select **IP Set and DHCP Information** by pressing the **Select** key in the Navigation key cluster or by pressing the number **2**.
- 4 Use the down arrow in the Navigation key cluster to scroll down the menu to **Firmware Version**.
- 5 Note the UNISlim Firmware version number and write it down.
 Compare the version number to the minimum-required UNISlim Firmware version (0625C39).
 UNISlim Firmware version names contain numbers and letters. Use the last three characters in a version to compare the version of UNISlim on an IP Phone 1140E (0625**C39**) with the minimum required version for the upgrade. Note that C23 is greater than **C39** and C1B is less than **C39**.

—End—

If the version number is equal to or higher than 0625C39, go to the section ["Convert UNISlim Firmware to SIP Firmware on the IP Phone 1140E"](#) (page 87).

If the number is lower than 0625C39, go to the section "[Upgrade UNISTim Firmware to the minimum required UNISTim Firmware](#)" (page 84) and follow the instructions to upgrade an IP Phone 1140E to the minimum-required version of UNISTim Firmware before you convert to SIP Firmware.

Upgrade UNISTim Firmware to the minimum required UNISTim Firmware

Use either of the following two methods to upgrade UNISTim Firmware.

1. UFTP download initiated by the server if the server supports this method of upgrading UNISTim Firmware. Refer to the appropriate documentation for your call server for instructions on using this method.
2. TFTP download on bootup.

If necessary, use the following procedure to configure the TFTP server.

Procedure 9

Configuring the TFTP server

Step	Action
1	The IP Phone 1140E always executes the TFTP download at bootup if a TFTP IP address is configured on the IP Phone 1140E after being initiated by the telephony call server.
2	Go to the TFTP server and create the 1140e.cfg provisioning file. The 1140e.cfg provisioning file is a clear text file. Create the provisioning file as shown in the next table.

Table 13

Sample 1140e.cfg provisioning file

```
[FW]
DOWNLOAD_MODE FORCED
VERSION 0625C23
FILENAME 0625C23.bin
```

This configuration file forces the firmware download of 0625C23.bin.

- 3 Download and copy the firmware to the TFTP server directory.
To download the UNISTim Firmware for the IP Phone 1140E from the Nortel Web site:
 1. Go to www.nortel.com/support.
 2. Log on to the Nortel Web site with a valid Nortel User ID and Password.

The **Technical Support** page appears.

3. Enter IP Phone 1140E in the **Knowledge and Solution Engine** box.
4. Select **SOFTWARE** in the **ALL TYPES** scroll down menu.
5. Press the gray arrow at the end of the **Knowledge and Solution Engine** box to obtain the **Search Results**.
6. From the **Search Results**, select the appropriate version of the UNISTim Firmware for the IP Phone 1140E, for example, **IP Phone 1140E Release 0625C23**
7. Place the selected firmware in the correct directory on the provisioning server.

- 4 In the IP Phone 1140E **Network Configuration** menu, change the **TFTP server address** and enter the correct TFTP server address.

This can be the provisioning server as defined in the chapter ["Configure the provisioning server" \(page 29\)](#).

- 5 Select the **Apply&Reset** Context-sensitive soft key to save the settings and reset the IP Phone 1140E.

The IP Phone 1140E downloads the firmware file. The display shows **[FW] reading...**

If the download is successful, the display shows **[FW] writing...** and the blue LED flashes.

After the firmware image is downloaded to the IP Phone 1140E, the display shows **[FW] finished...**, the blue LED stops flashing, and the IP Phone 1140E resets.

The IP Phone 1140E registers to the TPS with the new firmware version.

If the upgrade is unsuccessful, see the chapter ["Maintenance and troubleshooting" \(page 149\)](#) in the section **Download failures**.

—End—

Follow the next procedure to download the minimum required version of UNISTim Firmware automatically through TFTP on bootup.

Procedure 10**Downloading UNISlim Firmware automatically through TFTP on bootup****Step Action**

- 1 Press the **Globe/Services** key on the IP Phone 1140E twice quickly.
If the admin password prompt appears, enter the password **26567*738**
The Local Tools menu appears:

Table 14
Local Tools menu

- | |
|--|
| <ol style="list-style-type: none"> 1. Preferences 2. Local Diagnostics 3. Device Settings 4. Lock Menu |
|--|

- 2 Select **3. Device Settings** from the **Local Tools** menu.
The **Device Settings** screen appears.
- 3 If you are using DHCP, select Full or Partial. If you are manually configuring the IP address, netmask, and gateway address, select No.
- 4 If the DHCP option is set to Full or Partial, the IP address is automatically obtained.
- 5 Configure the TFTP IP address within the IP Phone 1140E Device Settings menu.

This can be the provisioning server as defined in the chapter ["Configure the provisioning server" \(page 29\)](#).
- 6 Select the **Apply&Reset** Context-sensitive soft key to save the settings and reset the IP Phone 1140E.

The IP Phone 1140E downloads the firmware file. The display shows **[FW] reading...**

If the download is successful, the display shows **[FW] writing...** and the blue LED flashes.

After the firmware image is downloaded to the IP Phone 1140E, the display shows **[FW] finished...** the blue LED stops flashing, and the IP Phone 1140E resets.

If the upgrade is unsuccessful, see the chapter "[Maintenance and troubleshooting](#)" (page 149) in the section **Download failures**.

—End—

Convert UNISlim Firmware to SIP Firmware on the IP Phone 1140E

The IP Phone 1140E can be ordered with UNISlim Firmware installed or with SIP Firmware installed. If an IP Phone 1140E is installed with UNISlim Firmware, it runs with SIP Firmware only if the firmware is converted from UNISlim to SIP. If the procedure to determine the UNISlim version number is complete, and, if necessary, the procedure to upgrade the UNISlim Firmware is complete, an IP Phone 1140E can be converted from UNISlim Firmware to SIP Firmware.

Compare the version number to the minimum required UNISlim Firmware version (0625C39).

UNISlim Firmware version names contain numbers and letters. Use the last three characters in a version to compare the version of UNISlim on an IP Phone 1140E (0625**C39**) with the minimum required version for the upgrade. Note that C23 is greater than **C39** and C1B is less than **C39**.

The conversion must be performed using TFTP.



WARNING

The TFTP download and upgrade of the Flash memory on the IP Phone 1140E may take a significant amount of time (possibly up to 10 minutes). Do not unplug or reboot the IP Phone 1140E during the process.

The next procedure explains how to download the SIP Firmware from the Nortel Web site.

Procedure 11

Downloading SIP Firmware for the IP Phone 1140E from the Nortel Web site

Step	Action
1	Go to www.nortel.com/support .
2	Log on to the Nortel Web site with a valid Nortel User ID and Password. The Technical Support page appears.
3	Enter IP Phone 1140E in the Knowledge and Solution Engine box.
4	Select SOFTWARE in the ALL TYPES scroll down menu.

- 5 Press the gray arrow at the end of the **Knowledge and Solution Engine** box to obtain the **Search Results**.
- 6 From the **Search Results**, select the appropriate version of the SIP Firmware for the IP Phone 1140E, for example, **IP Phone 1140E Release 0625C39D26.bin**.
- 7 Place the selected firmware on the provisioning server.

—End—

Follow the next procedure to convert the UNISTim Firmware to SIP Firmware on the IP Phone 1140E.

Procedure 12

Converting UNISTim Firmware to SIP Firmware using TFTP

Step	Action
1	Run the TFTP server (for example Tftpd32.exe).
2	Place firmware and configuration files in the folder of the TFTP server (for example 1140e.img F/W file and 1140e.cfg file) that contains the following lines:

Table 15
Sample 1140e.cfg configuration file

[FW] DOWNLOAD_MODE AUTO VERSION 06C25D26.bin FILENAME 1140e.img
--

- 3 Configure the IP Phone 1140E Device Settings TFTP IP address to the IP address where your TFTP server is running.

After you are finished the configuration, the IP Phone reboots and sends a request to the TFTP server.
- 4 Select the **Apply&Reset** Context-sensitive soft key to save the settings and reset the IP Phone 1140E.

The following messages display on the IP Phone 1140E as the IP Phone 1140E cycles through the conversion process, one after the other:
 1. [FW] Reading...
 2. [FW] Writing...

3. [FW] Finished...

The IP Phone 1140E then boots up with SIP Firmware.

If the conversion is unsuccessful, see the chapter "[Maintenance and troubleshooting](#)" (page 149).

- | |
|--|
| 1. TFTP file transfer takes approximately 15 seconds. |
| 2. File writing takes 2.5 minutes. The IP Phone 1140E displays the message [FW] writing... and the blue Data Waiting LED flashes. |
| 3. After the new SIP Firmware writing is finished, the blue LED stops flashing and the telephone displays [FW] finished and then reboots. |
| 4. The first time the SIP Firmware boots, the SIP Firmware performs a Flash File System conversion that takes 2.5 minutes. |

—End—

Voice Quality Monitoring

Feature overview

Proactive Voice Quality Monitoring (PVQMon or VQMon) allows the IP Phone 1140E with SIP Firmware to report voice quality statistics to a server in the network. The IP Phone 1140E with SIP Firmware collects various voice quality statistics, for example, packet loss, and sends the voice quality statistics to the server at regular intervals during a call. A subset of these statistics is also available for the user to view on the IP Phone 1140E by selecting the **Audio softkey** and then the **Monitor Audio Quality menu item**.

VQMon set-up

Configure the following parameters on the IP Phone 1140E with SIP Firmware to connect to the server and send the PVQMon statistics.

1. Enable the feature. To enable the feature, configure the VQMON_PUBLISH parameter in the device configuration file (see ["VQMon configuration commands" \(page 57\)](#)).
2. Configure the IP address of the PVQMon server. Configure the IP address of the PVQMon server in either of the following settings:
 - a. Set VQMON_PUBLISH_IP through the device configuration file (see ["VQMon configuration commands" \(page 57\)](#)).
 - b. Set PVQMon IP in Device Settings (see [Table 51 "PVQMon IP configuration" \(page 121\)](#))
3. Configure the remainder of the VQMon parameters in the device configuration file (see ["VQMon configuration commands" \(page 57\)](#)). These parameters provide threshold information to the IP Phone 1140E with SIP Firmware. A report is sent to the server when these thresholds are exceeded.

Server set-up

The IP Phone 1140E with SIP Firmware works with Telchemy server software. The name of the software is SQmediator and is available through Telchemy (www.telchemy.com). The minimum version required is release 1.0.

How VQMon works

The IP Phone 1140E with SIP Firmware gathers statistics about the current call when VQMon is enabled. Statistics are also gathered regarding the quality metrics of the current call. The call-related statistics contain condensed information about the SIP Session Description Protocol (SDP), the Call ID, the local and remote address, voice quality-related statistics, Zulu times for start-time and the time the report was sent.

The voice quality-related statistics include jitter, packet loss, delay, burst gap loss, listening R-factor, R-LQ, R-CQ, MOS-LQ and MOS-CQ. See [Table 16 "Glossary of RTCP XR metrics" \(page 92\)](#). More information on each of these metrics is provided in RFC3611 "RTP Control Protocol Extended Reports (RTCP XR)".

When the IP Phone 1140E detects that a particular voice quality metric has exceeded a threshold (defined in the device configuration file), the IP Phone sends a message to the server indicating that there is an issue. If the issue persists then the IP Phone reports another message indicating that there is an exceeded value at regular intervals. This happens continuously until the voice quality metric falls below the threshold value. As well, the IP Phone can send regular reports of the voice quality at time intervals defined in the device configuration file.

Table 16
Glossary of RTCP XR metrics

Burst	A period of high packet losses and / or discards. A burst is calculated in milliseconds.
Conversational R-factor	Voice quality metric based on burst packet loss and vocoder selection.
Delay	One way delay which includes end-to-end delay, jitter buffer delay and packetization delay. Delay is calculated in milliseconds.
Inter-arrival jitter	The variation in packet arrival times due to transmission (routing, queuing delay) through the network. Jitter is calculated in milliseconds.
Listening R-factor	Voice quality metric based on burst packet loss, transmission delay and burst loss.

MIU	Media Information Unit. MIU is a concept from VQMon. An MIU can be any size down to a 10 millisecond (8 sample) block. An MIU means a frame in the i200x implementation.
MOS	Mean Opinion Score. A subjective measurement of the voice quality of a voice call.
MOS_CQ	The VQMon conversational quality MOS score calculated for a call channel.
MOS_LQ	The VQMon listening quality MOS score calculated for a call channel.
Packet loss rate	The percentage of total packets loss versus packets received.
R-factor	A measurement of voice quality based on network impairments including burst packet loss, delay and encoding/decoding algorithm selection.

End of call report

The SIP phone sends a report using VQMON Publish message to the proxy. The proxy redirects the publish ID described within the report. End of call report is always generated if VQMON is enabled SIP phones do not negotiate or exchange messages with the device defined using PUBLISH_IP options.

Session interval report

The SIP phone can send voice quality reports at time intervals defined in the device configuration file. The minimum and default time interval is 60 seconds. If SIP phones send session interval reports more frequently, then a threshold violation has occurred.

Alert interval report

When a SIP phone detects that a voice quality metric has exceeded a threshold, the SIP phone initiates a timer which sends a message to the server every 5 seconds. When all voice quality metrics fall below the threshold values, the SIP phone stops sending VQMON Publish messages with the report. The alert interval report does not differ from the session interval reports or end of call reports.

Device Settings on the IP Phone 1140E with SIP Firmware

ATTENTION

An IP Phone 1140E with SIP Firmware displays different menus than the menus displayed on an IP Phone 1140E with UNISTim Firmware.

Introduction

This chapter details how to configure the **Device Settings** parameters on the IP Phone 1140E with SIP Firmware. This includes items such as the IP address, the subnet mask, and the gateway IP address of the IP Phone 1140E.

The **Device Settings** parameters are listed in the following table in the order they appear on the **Device Settings** menu of the IP Phone 1140E with SIP Firmware. Read the section at the end of this chapter that explains how to provision the **Device Settings** parameters. If you are familiar with the **Device Settings** parameters, skip the last section in the chapter and proceed to the provisioning instructions.

Table 17
Parameters in Device Settings menu

Enable 802.1x (EAP)
Device ID
Password
Enable 802.1ab (LLDP)
DHCP
No
Partial
Full
SET IP
Net Mask

Gateway	
DNS IP1	
DNS IP2	
Ntwk Port Speed	
	Auto
	10BT
	100BT
Ntwk Port Duplex	
	Auto
	Force Full
	Force Half
Disable Voice 802.1Q	
Voice VLAN	
	No VLAN
	DHCP
	LLDP VLAN Name
	LLDP MED
	Manual
VLAN Filter	
Ctrl Priority bits	
	Auto
	0 - 7
Media Priority bits	
	Auto
	0 - 7
Disable PC Port	
PC Port Speed	
	Auto
	10BT
	100BT
PC Port Duplex	
	Auto
	Force Full
	Force Half
Disable Data 802.1Q	

Data VLAN

No VLAN

LLDP VLAN NAME

Manual (value
from 1 to 4094)

Data Priority bits

Auto

0 - 7

PC-Port Untag all

Cached IP

Ignore GARP

Provisioning

Server URL

Protocol

TFTP

FTP

HTTP

Device ID

Password

PVQMon IP

NAT Traversal

NAT Signal

None

SIP Ping

STUN

NAT Media

None

STUN

NAT TTL

STUN S1 IP

STUN S2 IP

802.1x (EAP) Port-based network access control

Extensible Authentication Protocol (EAP) supports multiple authentication methods and represents a technology framework that facilitates the adoption of Authentication, Authorization, and Accounting (AAA) schemes, such as Remote Authentication Dial In User Service (RADIUS). RADIUS is defined in RFC 2865. The IP Phone 1140E with SIP Firmware supports only the MD5 authentication method.

802.1x defines the following three roles:

1. Supplicant—an IP Phone that requires access to the network to use network services.
2. Authenticator—the network entry point to which the supplicant physically connects (typically a Layer 2/3 switch). The authenticator acts as the proxy between the supplicant and the authentication server. The authenticator controls access to the network based on the authentication status of the supplicant.
3. Authentication server—performs authentication of the supplicant.

Enable and disable Network-level authentication through the EAP configuration menu.

Authorization

If 802.1x is configured and the IP Phone 1140E is physically connected to the network, the IP Phone 1140E (supplicant) initiates 802.1x authentication by contacting the Layer 2/3 switch (authenticator). The IP Phone 1140E also initiates 802.1x authentication after the Ethernet connection (network interface only) is restored following a network link failure.

However, if the IP Phone 1140E resets, it assumes the Layer 2 link has remained in service and is authenticated.

The IP Phone 1140E fails to authorize if the DeviceID and the IP Phone 1140E passwords do not match the DeviceID and IP Phone 1140E passwords provisioned on the RADIUS Server. The Layer 2 switch (authenticator) locks out the IP Phone 1140E and network access is denied. If this happens during reauthorization, all telephone services are lost. The connected PC operates as normal.

Device ID

The Device ID is for use with the 802.1x (EAP) protocol. If the 802.1x (EAP) is not used, then there is no prompt to enter the Device ID.

Password

The Password is for use with the 802.1x (EAP) protocol. If the 802.1x (EAP) is not used, there is no prompt to enter the Password.

802.1ab Link Layer Discovery Protocol

802.1ab Link Layer Discovery Protocol (LLDP) is a standard for discovering the physical topology between neighboring devices. 802.1ab LLDP defines a standard method for Ethernet network devices, such as switches, routers, and IP Phones to advertise information about themselves to other nodes on the network and to store the information they discover in a Management Information Base (MIB).

802.1ab (LLDP) takes advantage of the VLAN Name and Network Policy TLVs, and provides an automatic configuration of the phone network policy parameters. Key parameters, such as VLAN ID, L2 priority, and DSCP values are received from the switch and are automatically configured in the phone.

802.1ab Link Layer Discovery Protocol (LLDP) provides the following functionality

- Periodic transmission of advertisements containing device information, device capabilities and media specific configuration information to neighbors attached to the same network.
- Reception of LLDP advertisements from its neighbors.
- Implementation of behavioral requirements specified by Link Layer Discovery Protocol Media Endpoint Discovery (LLDP-MED).
- Storage of received data in local data structures, for example, in MIB modules.

TLVs

The information fields in each MIB are contained in a Link Layer Discovery Protocol Data Unit (LLDPDU) as a sequence of short, variable-length, information elements known as TLVs that each include type, length, and value fields. Each LLDPDU includes several mandatory TLVs plus optional TLVs. Optional TLVs may be inserted in any order.

The IP Phone supports both the transmit and receive LLDP mode.

Transmit direction

An LLDPDU transmitted by the IP Phone supports the following TLVs

1. Chassis ID
2. Port ID
3. Time To Live
4. End of LLDPDU
5. Port Description
6. System Description

7. System Capabilities
8. Port VLAN ID
9. Port And Protocol VLAN ID
10. VLAN Name
11. Protocol Identity
12. MAC/PHY Configuration Status
13. Power Via MDI
14. Link Aggregation
15. Maximum Frame Size
16. LLDP-MED Capabilities
17. Network Policy
18. Extended Power-via MDI
19. Inventory Firmware Revision
20. Inventory Manufacturer Name
21. Inventory Model Name

Receive direction

The IP Phone expects to receive the following TLVs

1. Chassis ID
2. Port ID
3. Time To Live
4. End of LLDPDU
5. System Capabilities
6. VLAN Name
7. MAC/PHY Configuration Status
8. LLDP-MED Capabilities
9. Network Policy
10. Location Identification

Table 18
TLV formats

TLV	Fields
Chassis ID	Length = 6 Chassis Subtype = 5 [IP Address] Chassis ID = IP Phone IP Address

TLV	Fields
Port ID	Length = 7 Port Subtype = 3 [MAC Address] Port ID = IP Phone MAC address
Time To Live	Length = 2 TTL= 180 [seconds]
End Of LLDPDU	Length = 0
Port Description	Length = 15 Port Description = "Nortel IP Phone"
System Description	Length = the length of the system description string, System Description = "Nortel IP Telephone, xxx, Firmware: 0604D97" where: xxx = 2001, 2002, 2004, 2007, 1110, 1120E, 1140E, 1150E Firmware = firmware version. "0604D97" is an example only.
System Capabilities	Length = 4 System capabilities = 0x24 [Telephone + Bridge] Enabled capabilities = 0x24 If you disable the PC Ethernet port, the advertised enabled capabilities configured to Telephone only.
Port VLAN ID	PVID = 0 The phone does not support port-based VLAN operation.
Port And Protocol VLAN ID	PPVID = 0 Port and Protocol VLAN is not supported and not enabled.
VLAN Name	VLAN name field is configured to "data" and "voice".
Protocol Identity	<ol style="list-style-type: none"> STP: Protocol identity = the first 8 bytes of an STP PDU starting with the Ethertype field. Length = 8 Protocol Identity = 0x00 0x26 (type/length field of Ethernet packet, size=38) 0x42 0x42 0x03 (LLC header indicating STP) 0x00 0x00 (Protocol Identity field from STP BPDU) 0x00 802.1x: Length = 3 Protocol identity = 0x888E—(802.1x Ethertype) 0x01—(Version field from 802.1x frame) LLDP: Length = 2 Protocol identity = 0x88CC—(LLDP Ethertype)
MAC/PHY Configuration/Status	Auto-negotiation support/status = Bit 0 = 1 [Auto-negotiation supported] Bit 1 = 1 or 0, depending on the current auto-negotiation status, for example, either enabled or disabled.

TLV	Fields
	PMD auto-negotiation advertised capability = 0x4000 - 10BASE-T half duplex mode 0x2000 - 10BASE-T full duplex mode 0x0800 - 100BASE-TX half duplex mode 0x0400 - 100BASE-TX full duplex mode 0x0002 - 1000BASE-TX half duplex mode 0x0001 - 1000BASE-TX full duplex mode Operational MAU Type = 10 – UTP MAU, 10BT, half duplex mode 11 – UTP MAU, 10BT, full duplex mode 15 - 2-pair Category 5 (CAT5) UTP, 100BT, half duplex mode 16 - 2-pair CAT5 UTP, 100BT, full duplex mode 29 – 4-pair CAT5 UTP, 1000BT, half duplex mode 30 – 4-pair CAT5 UTP, 1000BT, full duplex mode
Power Via MDI	MDI power support = 0: Bit 0 = 0 – Powered Device Bit 1 = 0 – PSE MDI power not supported Bit 2 = 0 – PSE MDI power state disabled Bit 3 = 0 – PSE pair selection can not be controlled PSE power pair = 1 Power Class = 2 for 2001/2002/2004/1110 phones Power Class = 3 or 2007/1120E/1140E/1150E phones
Link Aggregation	Aggregation status = 0; the link is not capable of being aggregated, and currently is not in aggregation. Aggregated Port ID = 0
Maximum frame size	The MAC/PHY supports an extension of the basic MAC frame format for Tagged MAC frames. The maximum frame size is configured to 1522.
LLDP-MED System Capabilities	Bit 0 = 1—LLDP-MED Capabilities—supported Bit 1 = 1—Network Policy—supported Bit 2 = 1—Location Identification—supported Bit 3 = 0—Extended Power using MDI-PSE—not supported Bit 4 = 1—Extended Power using MDI-PD—supported Bit 5 = 1—Inventory—supported The Class Type field can be configured to 3 -Telephone
Network Policy Discovery	Application Type-1—voice Unknown Policy Flag (U)—1 only if the policy is unknown Tagged Flag (T)—configure accordingly Reserved (X)-0 VLAN ID—set accordingly L2 Priority—set accordingly DSCP Value—set accordingly

TLV	Fields
Location Identification Discovery	<p>Coordinate-based LCI—16 bytes</p> <p>Civic Address LCI I—variable length</p> <p>This format can have more than one address element and one address element can range from a minimum of 7 to 256 bytes.</p> <p>ECS ELIN I—variable between 10 and 25 bytes</p> <p>Although location is received, it is not available to end user in this release of the SIP firmware.</p>
Extended Power-via MDI Discovery	<p>Power Type = 01—PD Device</p> <p>Power Source = 00—Unknown.</p> <p>There is no hardware support for determining the power source.</p> <p>Power Priority = 0010—High</p> <p>Power Value = Maximum power required as shown below:</p> <p>2001 NTDU90 = 4</p> <p>2002 NTDU91 = 4.5</p> <p>2004 NTDU92 = 4.5</p> <p>2007 NTDU96 = 10</p> <p>1110 NTYS02 = 5</p> <p>1120E NTYS03 = 8</p> <p>1140E NTYS05 = 8</p> <p>1150E = 7.6</p>
Firmware Revision	Configure to the firmware version being used, for example, 0604D97.
Manufacturer Name	<p>“Nortel-xy”,</p> <p>where: xy is a 2-digit manufacturer code as shown below:</p> <p>2001: Code 01</p> <p>2002: Code 01</p> <p>2004: Code 01</p> <p>2007: Code 01</p> <p>2033: Code 03</p> <p>1210: Code 04</p> <p>1220: Code 04</p> <p>1230: Code 04</p> <p>1110: Code 01</p> <p>1120: Code 01</p> <p>1140: Code 01</p> <p>1150: Code 02</p>
Model Name	Contains a string, which specifies the phone model, for example, “IP Phone xxx”, where, xxx is one of the following values: 2001, 2002, 2004, 2007, 1110, 1120E, 1140E, 1150E.

DHCP

There are three methods of provisioning DHCP.

1. **No DHCP (Manual configuration):** All the Device Settings parameters are configured manually on the IP Phone 1140E with SIP Firmware.
2. **Partial DHCP:** The IP Phone 1140E with SIP Firmware is configured to get a standard set of Device Settings parameters from the DHCP server.
3. **Full DHCP:** The IP Phone 1140E with SIP Firmware is configured to get as many Device Settings parameters as possible from the DHCP server.

NO DHCP mode

No DHCP mode is also known as Manual Device Settings. In this mode, the IP Phone 1140E does not need a DHCP server because no DHCP requests are sent during startup. All necessary parameters must be configured manually or through the device configuration file.

The minimum following parameters must be configured to achieve normal operation:

- SET IP
- Net Mask
- Gateway
- DNS IP
- Provisioning server IP, protocol, and, if user authentication is required to access the provisioning server, the user credentials

Note: TFTP protocol does not require user authentication. Device ID and password are ignored. FTP protocol requires user authentication and the default Device ID is anonymous with no password.

HTTP protocol can operate with or without user authentication. If no authentication is required, make sure to clear the Device ID and password fields in the configuration dialogue.

SET IP

Select SET IP in the Device Settings menu to configure the IP address.

Net Mask

Enter a subnet mask In the Net Mask field.

Gateway

Enter an IP address of the local gateway in the Gateway field.

DNS IP1 and DNS IP2

Configure the Domain Name Servers (DNS) IP addresses DNS IP1 and DNS IP2.

Ntwk Port Speed

There are three options to provision the network port speed.

- Auto. Link speed is auto negotiated with the network device and attached PC.

Note: 1000BT must be negotiated using auto.

- 10BT. Link speed is available for up to 10 Megabit Full Duplex on the network and the PC port.
- 100BT. Link speed is available for up to 100 Megabit Full Duplex on the network and the PC port.

Ntwk Port Duplex

There are three options available to provision the Network Port Duplex for Network Port Speed of 10BT or 100BT.

- Auto. Duplex is auto negotiated. It is recommended that Auto Negotiate mode is used on the network and the IP Phone.
- Force Full. Duplex is forced to FULL. Use Force Full mode only when the network is forced Full Duplex. Otherwise a duplex mismatch will result.
- Force Half. Duplex is forced to HALF. Use Force Half mode only when the network is forced Half Duplex. Otherwise a duplex mismatch will result.

Disable Voice 802.1Q

If 802.1Q is disabled, standard Ethernet frames are transmitted. If 802.1Q is enabled, all frames transmitted by the Ethernet driver have the 802.1Q tag bytes inserted between the source MAC address and the protocol type field.

Voice VLAN

Configure the Voice VLAN.

Table 19
Telephony Port (incoming)

Voice VLAN Setting	Result
No VLAN (default setting)	All telephony traffic that is transmitted on the telephony port is forwarded untagged.

DHCP	<p>Enable Automatic VLAN Discovery for Voice VLAN. Initial DHCP request is forwarded based on the Data VLAN policy (for example, untagged if Data VLAN is disabled, or tagged with the Data VLAN ID if Data VLAN is enabled) to obtain Voice VLAN IDs. Subsequent DHCP requests using the acquired Voice VLAN IDs and the resolution of a Voice VLAN determine the final Voice VLAN ID used for signaling and the media traffic on the telephony port. Note: This means that if a Data VLAN ID is configured in the IP Phone 1140E and the Voice VLAN is set to DHCP (Automatic Voice VLAN Discovery), the DHCP server to provide Voice VLAN information must be on the Data VLAN. If the DHCP server is not on the Data VLAN, the IP Phone 1140E fails to receive the DHCP information and fails to register. In addition, another DHCP server must reside in the Voice VLAN to provide the network parameters required. For more information, see the section on DHCP Auto VLAN Discovery. DHCP does not appear if it is disabled.</p>
LLDP MED	<p>If LLDP-MED is configured and the switch supports the Network Policy TLV, Voice VLAN is configured based on the data received from the switch in the Network Policy TLV. LLDP-MED does not appear if Voice VLAN is not enabled.</p>
LLDP VLAN NAME	<p>If LLDP VLAN Name is configured and the switch is provisioned with the "data" VLAN Name TLV, data VLAN is configured based on the VLAN Id value received from the switch in the "data" VLAN Name TLV.</p> <p>When data VLAN is set to Auto, and the switch is provisioned with the "data" VLAN Name TLV, data VLAN is configured based on the VLAN Id value received from the switch in the "data" VLAN Name TLV.</p> <p>LLDP VLAN Name does not appear if Voice VLAN is not enabled.</p>
Manual (value from 1 to 4094)	<p>All telephony traffic that is transmitted on the telephony port has an 802.1q header appended and its Voice VLAN ID is set to the value manually configured here.</p> <p>Enter a value from 1 to 4094.</p>

VLAN Filter

Configure the VLAN Filter (not available if Voice VLAN is set to No).

Table 20
VLAN Filter

VLAN Filter Setting	Result
Enabled (box is checked)	Traffic is forwarded to the IP Phone 1140E port, based on a review of the MAC address of the packet as well as the 802.1q tag value. Traffic is forwarded through the IP Phone 1140E port, (to the network stack of the telephone), only if the packet matches the MAC address of the IP Phone 1140E and contains the Voice VLAN tag. If the Automatic VLAN Discovery feature is used, the filter is adjusted dynamically as the IP Phone 1140E validates the VLAN suggested by DHCP.
Disabled	The VLAN filter is disabled by default. If the VLAN filter is disabled, traffic is forwarded to the IP Phone 1140E port based only on a review of the MAC address of the packet. The IP Phone 1140E accepts traffic addressed to the MAC address of the IP Phone 1140E as well as any broadcast or multicast packets from any VLAN.

Ctrl Priority bits

802.1Q priority bits for the control or signaling stream. There is not a control priority bit prompt if 802.1Q is not enabled.

Media Priority bits

802.1Q priority bits for the media (audio) stream. There is not a media priority bit prompt if 802.1Q is not enabled.

Disable PC Port

With a disabled PC Port, the IP Phone 1140E with SIP Firmware does not receive or send packets to or from the PC port. No device can use the PC port to connect to the network.

Data VLAN

Configure the Data VLAN (the VLAN applicable to the PC port). The behavior of the PC port is summarized in the following table.

Table 21
PC Port (incoming traffic from PC)

Data VLAN	Result
No VLAN	All traffic received on the PC port is forwarded based on the MAC address. The packets are not modified in any way.

LLDP VLAN NAME	If Data VLAN is configured and the switch is provisioned with the "data" VLAN Name TLV, Data VLAN is configured based on the VLAN ID value received from the switch in the "data" VLAN Name TLV.
Manual (value from 1 to 4094)	All untagged packets received from the PC have an 802.1q header appended and the VLAN ID is set to the value that is manually provisioned in this field. Any packet arriving on the PC port that is already tagged with matching VLAN ID is forwarded as is. Packets tagged with different VLAN IDs are dropped. Enter a value from 1 to 4094.

Table 22
PC Port (outgoing)

Data VLAN	Result
No	All traffic received on the network port and telephony port is forwarded to the PC port based on MAC address only. The packets are not modified in any way.
4095 (Note that the value 4095 is not a valid VLAN ID. This is intentional to ensure a proper VLAN ID is entered.)	Traffic is forwarded to the PC port based on a review of the MAC address of the packet, as well as the value configured in the Data VLAN field. Traffic is forwarded out the PC port (to the PC) only if the packets contain the Data VLAN tag. Untagged traffic and traffic with a VLAN tag other than the Data VLAN are dropped.

Data Priority bits

There is not a prompt for data priority bits if 802.1Q is not enabled.

PC-Port Untag all

Configure PC-Port Untag-All, which configures PC-Port VLAN Tag Stripping.

When enabled, all outgoing traffic to the PC has the VLAN tag removed. When disabled, packets are sent to the PC as is if Data VLAN is set to No or if Data VLAN is enabled and the VLAN tag matched. Packets with different VLAN ID are discarded.

Cached IP

Leave unchecked to conform to the DHCP standard and to obtain an IP address from the DHCP server. Only check Cached IP to force the IP Phone to start with a cached IP address in the event that the IP Phone cannot connect to the DHCP server and obtain an IP address.

Port Speed and Duplex

In the Network menu, Auto Negotiation mode is the default setting for initial startup. Typically, the IP Phone 1140E is connected to a network that supports Auto Negotiation, and the IP Phone 1140E selects the best speed and duplex mode available. There is no intervention required under normal operation.

ATTENTION

Nortel recommends that Auto Negotiation mode is used on the network and the IP Phone 1140E. Use Full Duplex mode only when the entire network is running in Full Duplex mode. Otherwise, a duplex mismatch results.

If the IP Phone is connected to a network configured for Full Duplex mode only, the IP Phone 1140E cannot automatically negotiate the proper configuration. Therefore, in this instance, to allow the IP Phone 1140E to work at the optimum speed and duplex mode, Full Duplex mode must be enabled.

Ignore GARP

Gratuitous Address Resolution Protocol (GARP) Protection prevents the IP Phone 1140E with SIP Firmware from GARP spoof attacks on the network.

The IP Phone 1140E with SIP Firmware provides the ability to ignore GARP messages.

Provisioning

The next four menu items apply to configuring the provisioning parameters.

Server URL Enter the numeric IP address, name, or URL of the provisioning server in the Server URL box.

Protocol Select the protocol used to access the provisioning server, either TFTP, FTP, or HTTP.

Device ID Enter the Device ID (User ID) used by the provisioning server for authentication of the IP Phone 1140E with SIP Firmware.

Password Enter the password used by the provisioning server for authentication of the IP Phone 1140E with SIP Firmware.

PVQMon IP

If the Proactive Voice Quality Monitor (PVQMon) server is available on the network, enter the PVQMon IP address.

NAT Traversal

The next five menu items apply to configuring the NAT Traversal Method if the IP Phone 1140E with SIP Firmware is connected to the network through a Network Address Translation device or a firewall.

NAT Signal The IP Phone 1140E with SIP Firmware supports two methods of NAT traversal of signaling path: SIP_PING and STUN.

The default NAT traversal method is None.

SIP_PING is a legacy Nortel Proprietary protocol for NAT Traversal for SIP signaling only. STUN is an Internet standard for NAT traversal.

If the value for NAT traversal is not set to None, this parameter overrides the value of the parameter SIP_PING specified by the device configuration file for NAT_SIGNALLING.

If the value for NAT traversal is set to None, the value of SIP_PING, if specified by the device configuration file for NAT_SIGNALLING, is used instead.

NAT_SIGNALLING is required for networks that use STUN or SIP_PING for NAT traversal.

NAT Media The IP Phone 1140E with SIP Firmware supports STUN for media path the NAT traversal. The NAT Media feature can be disabled by setting the NAT_Media field in **Device Settings** menu to NONE.

Note: STUN protocol cannot coexist with Application Layer Gateway (ALG), Media Portals, or RTP Proxy servers. If STUN is selected, ensure none of these devices are configured in the SIP Proxy server.

NAT TTL Enter the Time to Live (TTL) in seconds for the NATport if STUN is enabled. The Nortel IP Phone 1140E with SIP Firmware pings the open ports on an interval less than the TTL to prevent the NAT from tearing down the ports.

STUN S1 IP Enter an IP address for the STUN S1 IP device. The STUN server must reside in the public Internet for STUN protocol to be effective.

STUN S2 IP Enter an IP address for the STUN S2 IP device. The STUN server must reside in the public Internet for STUN protocol to be effective.

Configure the device settings

This section describes how to provision the **Device Settings** parameters regardless of DHCP mode (Full DHCP, Partial DHCP, or No DHCP).

Parameters that are not needed in Full or Partial DHCP mode are grayed out on the IP Phone 1140E screen and cannot be modified.

Follow the next procedure to provision the **Device Settings** parameters on the IP Phone 1140E with SIP Firmware.

Procedure 13

Provisioning the Device Settings parameters

Step	Action
------	--------

- | | |
|---|--|
| 1 | Press the Globe/Services key on the IP Phone 1140E quickly twice. The Network menu appears: |
|---|--|

Table 23
Network menu

- | |
|--------------------|
| 1. Server Settings |
| 2. Device Settings |
| 3. Diagnostics |
| 4. Lock |

- | | |
|---|--|
| 2 | To navigate in the Network menu, use the Navigation key cluster : |
|---|--|

Table 24

Using the Navigation key cluster to navigate in the Network menu

Key	Action
Down	Moves highlight down
Up	Moves highlight up
Right	Selected current menu item
Left	Closes menu
Select key (center of cluster)	Selects current menu item

Menu items can also be accessed through the IP Phone 1140E keypad. Press the number (1-4) corresponding to a menu item to highlight that menu item. Press the **Select** key to select the highlighted menu item.

To close this menu, use the **Quit** key.

- | | |
|---|--|
| 3 | In the Network, choose 2. Device Settings . |
| 4 | Enter the admin password. Press the OK Context-sensitive soft key. |

- 5 Use the **Navigation key cluster** to edit an item:

Table 25

Using the Navigation key cluster to navigate in the Device Settings menu

Key	Action
Down	Opens list or moves highlight down list.
Up	Moves highlight up list.
Right	Selects the next item or moves the cursor right in an edit item.
Left	Selects the previous item or deletes a character in the edit field.
Select key (center of cluster)	Selects the highlighted item in the combo box or ends edit mode.

In Edit mode, the first field of the item is highlighted. Press the **Apply&Reset Context-sensitive soft key** to save settings in the Device Settings menu after all the necessary changes are made. This action resets the IP Phone 1140E.

Press the **Return Context-sensitive soft key** to exit the Device Settings menu without saving any changes.

- 6 Navigate to the first item in the Device Settings menu to configure **EAP**.

Table 26

EAP configuration

Enable 802.1x (EAP): <input type="checkbox"/> Device ID: Password:
--

A check mark appears in the check box if the item is active.

- 7 If the **Enable 802.1x (EAP)** check box is checked, fill in the **Device ID** and the **Password**.

- 8 Configure 802.1ab Link Layer Discovery Protocol Data (LLDP) in the Enable 802.1ab (LLDP) menu.

Use the right arrow in the **Navigation key cluster** to highlight the Enable 802.1ab (LLDP) box.

A check mark appears in the check box if the item is active.

Table 27
802.1ab configuration

Enable 802.1ab (LLDP): <input type="checkbox"/>

- 9** Configure DHCP in the Device Settings menu.
- Use the right arrow in the **Navigation key cluster** to highlight the DHCP box. Press the **Select** key to access a list of DHCP mode choices:
- No
 - Partial
 - Full
- Scroll through the list with the Up/Down arrows and select the appropriate parameter. Press the **Select** key to select No.

Table 28
DHCP configuration

DHCP :	No	▼	
	No		
	Partial		
	Full		

- 10** If DHCP is disabled, select SET IP in the Device Settings menu to configure the IP address.

Table 29
SET IP configuration

SET IP : 0.0.0.0

- 11** If DHCP is disabled, configure Net Mask and Gateway. Enter a subnet mask in the Net Mask field and enter the IP address of the local gateway in the Gateway field.

Table 30
Net Mask and Gateway configuration

Net Mask : 0.0.0.0
Gateway : 0.0.0.0

- 12** If DHCP is not Full DHCP, to configure the DNS enter the IP address of the local DNS servers DNS IP1 and DNS IP2.

Table 31
DNS configuration

DNS IP1 : 0.0.0.0 DNS IP2 : 0.0.0.0
--

13 Configure Ntwk Port Speed in the Device Settings menu.

Use the right arrow in the **Navigation key cluster** to highlight the Ntwk Port Speed box. Press the **Select** key to access a list of Ntwk Port Speed choices:

- Auto
- 10BT
- 100BT

Scroll through the list with the Up/Down arrows and select the appropriate parameter. Press the **Select** key to select a Ntwk Port Speed.

Table 32
Ntwk Port Speed configuration

Ntwk Port Speed :	Auto	▼	
	Auto		
	10BT		
	100BT		

14 Configure Ntwk Port Duplex in the Device Settings menu.

Use the right arrow in the **Navigation key cluster** to highlight the Ntwk Port Duplex box. Press the **Select** key to access a list of Ntwk Port Duplex choices:

- Auto
- Force Full
- Force Half

Scroll through the list with the Up/Down arrows and select the appropriate parameter. Press the **Select** key to select Ntwk Port Duplex.

Table 33
Ntwk Port Duplex configuration

Ntwk Port Duplex :	Auto	▼	
	Auto		
	Force Full		
	Force Half		

- 15** Disable Voice 802.1Q . Press the **Select** key to disable Voice 802.1Q.
A check mark in the Disable Voice 802.1Q box indicates that Voice 802.1Q is disabled.

Table 34
Voice 802.1Q configuration

Disable Voice 802.1Q: <input type="checkbox"/>
--

- 16** Configure Voice VLAN.
- Use the right arrow in the **Navigation key cluster** to highlight the Voice VLAN box. Press the **Select** key to access a list of Voice VLAN mode choices.
- Scroll through the list with the Up/Down arrows and select the appropriate parameter. Press the **Select** key to select the parameter.

Voice VLAN:	LLDP VLAN NAME	▼	
	No VLAN		
	DHCP		
	LLDP MED		
	LLDP VLAN NAME		
	Value from 1 to 4094		

If 802.1ab (LLDP) or DHCP are not enabled, DHCP, LLDP MED, and LLDP VLAN do not appear in the list of Voice VLAN mode choices.

- 17** Enable or disable the VLAN Filter. Press the **Select** key to enable the VLAN Filter.
- A check mark in the VLAN Filter box indicates that the VLAN Filter is enabled.

Table 35
VLAN Filter configuration

VLAN Filter : <input type="checkbox"/>
--

- 18** If 802.1Q is enabled, configure Ctrl Priority bits.

Use the right arrow in the **Navigation key cluster** to highlight the Ctrl Priority bits box. Press the **Select** key to access a list of Ctrl Priority bits choices.

Scroll through the list with the Up/Down arrows and select the appropriate parameter. Press the **Select** key to select the parameter.

Table 36
Ctrl Priority bits configuration

Ctrl Priority bits :	Auto	▼	
	Auto		
	Value from 0 to 7		

- 19** If 802.1Q is enabled, configure Media Priority bits.

Use the right arrow in the **Navigation key cluster** to highlight the Media Priority bits box. Press the **Select** key to access a list of Media Priority bits choices.

Scroll through the list with the Up/Down arrows and select the appropriate parameter. Press the **Select** key to select the parameter.

Table 37
Media Priority bits configuration

Media Priority bits :	Auto	▼	
	Auto		
	Value from 0 to 7		

- 20** Disable PC Port configuration. Press the **Select** key to turn off the PC port on the IP Phone 1140E.

A check mark in the Disable PC Port box indicates the PC port is disabled.

Disable PC Port can be used in an environment where administrators do not want users accessing the data network through the built-in PC port.

Table 38
Disable PC Port configuration

Disable PC Port : <input type="checkbox"/>
--

- 21** If the PC port is not disabled, configure PC Port Speed in the Device Settings menu.

Use the right arrow in the **Navigation key cluster** to highlight the PC Port Speed box. Press the **Select** key to access a list of PC Port Speed choices:

- Auto
- 10BT
- 100BT

Scroll through the list with the Up/Down arrows and select the appropriate parameter. Press the **Select** key to select a PC Port Speed.

Table 39
PC Port Speed configuration

PC Port Speed :	Auto	▼	
	Auto		
	10BT		
	100BT		

- 22** If the PC port is not disabled, configure PC Port Duplex in the Device Settings menu.

Use the right arrow in the **Navigation key cluster** to highlight the PC Port Duplex box. Press the **Select** key to access a list of PC Port Duplex choices:

- Auto
- Force Full
- Force Half

Scroll through the list with the Up/Down arrows and select the appropriate parameter. Press the **Select** key to select PC Port Duplex.

Table 40
PC Port Duplex configuration

PC Port Duplex :	Auto	▼	
	Auto		
	Force Full		
	Force Half		

- 23** Disable Data 802.1Q . Press the **Select** key to disable Data 802.1Q.

A check mark in the Disable Data 802.1Q box indicates that Data 802.1Q is disabled.

Table 41
Data 802.1Q configuration

Disable Data 802.1Q: <input type="checkbox"/>

24 Configure the Data VLAN.

It is possible to Enable or disable separate VLAN for anything other than voice and signaling.

Use the right arrow in the **Navigation key cluster** to highlight the Data VLAN box. Press the **Select** key to access a list of Data VLAN mode choices:

Scroll through the list with the Up/Down arrows and select the appropriate parameter. Press the **Select** key to select the parameter.

Table 42
Data VLAN Configuration in Device Settings menu

Data VLAN :	LLDP VLAN NAME	▼	
	No VLAN		
	LLDP VLAN NAME		
	Value from 1 to 4094		

If 802.1ab (LLDP) is disabled, Data VLAN cannot be automatically configured. LLDP VLAN NAME does not appear in the in the list of Data VLAN mode choices.

25 If 802.1Q is enabled, configure Data Priority bits.

Use the right arrow in the **Navigation key cluster** to highlight the Data Priority bits box. Press the **Select** key to access a list of Data Priority bits choices:

Scroll through the list with the Up/Down arrows and select the appropriate parameter. Press the **Select** key to select the parameter.

Table 43
Data Priority bits configuration

Data Priority bits :	Auto	▼	
	Auto		
	Value from 0 to 7		

- 26** Configure PC-Port Untag-All, which configures PC-Port VLAN Tag Stripping. Check the check box by pressing the **Select** key in the Navigation key cluster to enable this item.
- If the check box is not checked, tag stripping is disabled and the packet is sent to the PC port unmodified. If the check box is checked, tag stripping is enabled and the 802.1q header is removed (assuming one exists) from the packet before it is forwarded through the PC port.
- If Data VLAN is enabled in step 15, PC-Port Untag All is enabled and the check box is checked by default. In this case, the outgoing tag is stripped. Override the default by pressing the **Select** key to remove the checkmark.
- If Data VLAN is disabled, PC-Port Untag All is disabled and the check box is not checked by default. In this case, the ingress tag is not stripped. The default can be overridden by pressing the **Select** key, which places a checkmark in the box.

Table 44
PC-Port Untag-All Configuration in Device Settings menu

PC-Port Untag all : <input type="checkbox"/>
--

- 27** Configure Cached IP.
- Leave unchecked to conform to the DHCP standard and to obtain an IP address from the DHCP server. Only check Cached IP to force the IP Phone to start with a cached IP address in the event that the IP Phone cannot connect to the DHCP server and obtain an IP address.
- To force the IP Phone to start with a cached IP address, press the **Select** key.
- A check mark in the Cached IP box indicates that the IP Phone is forced to start with a cached IP address.

Table 45
Cached IP configuration

Cached IP: <input type="checkbox"/>

- 28** Configure the Ethernet port mode.
- Ignore GARP: GARP requests handling. The default can be overridden by pressing the **Select** key, which places a check mark in the box.

Table 46
Ethernet port mode configuration

Ignore GARP: ☐

- 29** Enter Provisioning Parameters: Server URL, Protocol, Device ID, and Password.

The next four tables show menu items in the Device Settings menu that apply to configuring the Provisioning parameters.

- **Server URL:** Enter the URL of the provisioning server in the **Server URL** box.

Table 47
Server URL configuration

Provisioning

Server URL:

0.0.0.0

- **Protocol:** Select either TFTP, FTP, or HTTP used for provisioning in the **Protocol** box.

Table 48
Protocol configuration

Protocol:

TFTP

TFTP

FTP

HTTP

- **Device ID:** Enter the **Device ID** used by the provisioning server for authentication of the IP Phone 1140E.

Note 1: TFTP does not require a Device ID. If FTP or HTTP are used, enter the information for a Device ID

Note 2: Enter your User ID as the Device ID.

Table 49
Device ID configuration

Device ID :

XXXXXXXXXX

- **Password:** Enter the **Password** used by the provisioning server for authentication of the IP Phone 1140E. Note: TFTP does not require a Password. FTP and HTTP do require a password.

Table 50
Password configuration

Password :	XXXXXXXXXX
------------	------------

30 Configure the PVQMon IP address.

If this server is available on the network, enter the PVQMon IP address.

Table 51
PVQMon IP configuration

PVQMon IP :	0. 0. 0. 0
-------------	------------

31 Configure the NAT Traversal Method. Use one of the following methods to traverse Network Address Translation (NAT) devices for the IP Phone 1140E with SIP Firmware.

The next five tables show menu items in the Device Settings menu that apply to configuring the NAT Traversal Method.

- **NAT Signal**

Select either None, SIP_PING, or STUN to define the NAT traversal mode for SIP signaling.

Table 52
NAT Signal configuration

NAT Traversal					
NAT Signal :	<table border="1" style="width: 100%;"> <tr><td>STUN</td></tr> <tr><td>None</td></tr> <tr><td>SIP Ping</td></tr> <tr style="background-color: #cccccc;"><td>STUN</td></tr> </table>	STUN	None	SIP Ping	STUN
STUN					
None					
SIP Ping					
STUN					

- **NAT Media**

Select either None or STUN to define NAT traversal mode for signaling.

Table 53
NAT media configuration

NAT Media :	STUN
	None
	STUN

- **NAT TTL:**

Enter the Time to Live (TTL) in seconds for the NAT traversal.

Table 54
NAT TTL configuration

NAT TTL :	000
-----------	-----

- **STUN S1 IP:** Enter an IP address for the STUN S1 IP device.

Table 55
STUN S1 IP configuration

STUN S1 IP :	00. 00. 00. 00
--------------	----------------

- **STUN S2 IP:**

Enter an IP address for the STUN S2 IP device.

Table 56
STUN S2 IP configuration

STUN S2 IP :	00. 00. 00. 00
--------------	----------------

—End—

Multiple Appearance Directory Number (Single Call Arrangement)

Note: Only users of the Communication Server 2000 (CS 2000) and Communication Server 2100 (CS 2100) have access to the Multiple Appearance Directory Number (MADN) Single Call Arrangement (SCA) feature at this time.

The Multiple Appearance Directory Number (MADN) allows a Directory Number (DN) to appear on more than one IP Phone 1140E with SIP Firmware. The MADN with Single Call Arrangement (SCA) feature allows multiple telephones to appear as a single line to a caller. Any one of the IP Phone 1140E telephones in a group with MADN can initiate or answer a call, but only one call can be active at any given time. Any other user in the group can join the active call by picking up the handset of the IP Phone 1140E with SIP Firmware.

With the MADN SCA feature configured on multiple telephones of different registered SIP users, the telephones share one single DN. An incoming call to this DN causes all the telephones in the group to ring.

Any user of an IP Phone 1140E with the MADN SCA feature can put a call on hold or can prevent others from joining in the active call.

If a user's group is active (as seen by line icon being off-hook) and the user picks up the handset, the user is automatically joined to an ongoing MADN call (unless the server restricts this feature for privacy or other factors).

Vertical services

Vertical services are CS 2000 and CS 2100 features that can be activated or deactivated by dialing a defined code, for example, Privacy. Even though no more than one active session can be established for the MADN SCA group, members of the group can still enter certain vertical services.

Currently, the available vertical service is Privacy.

Privacy

A user can activate the privacy service by putting the current session on hold and dialing the privacy code. The CS 2000 connects the IP Phone 1140E to the Media Application Server (MAS) to hear a confirmation for its request and terminates the session. The user takes the original session off hold.

Privacy access codes

The privacy access codes are: PRV, PRLA, PRLC.

For example: PRV = 191 PRLA = 192 PRLC = 193

If the initial state of the MADN group is nonprivate, the PRV access code is used to toggle between privacy on and privacy off. If the initial state of the MADN group is private, the PRLA access code allows bridging and PRLC closes it.

Feature dependencies and restrictions

The minimum release to support MADN SCA feature is 1.1. Multi-User login is not supported.

Emergency 911 - Operator control of disconnect

Overview

For Emergency 911 services to be effective, it is vital that a system maintain a connection with the 911 operator until the user provides all necessary information. In traditional 911 telephony, if a 911 call is disconnected before the necessary information is provided, mechanisms are in place to allow the 911 operator to ring back the calling party. In VoIP calls, 911 support is an evolving standard.

If the user of an IP Phone 1140E attempts to disconnect a 911 call after the 911 operator has answered, the IP Phone 1140E switches to hands-free mode. Once the IP Phone 1140E is in hands-free mode, the 911 call can only be disconnected by the operator.

Emergency 911 calls originate on the IP Phone 1140E and are completed by the Call Server. The Call Server communicates with the 911 network or 911 systems for routing, call control, and location information. Although the IP Phone 1140E allows users to enter location information, this location information is not used by all Call Servers. Some Call Servers derive the location information based on the number and location databases. Characteristics of Emergency 911 calls and limitations of Emergency 911 calls using the IP Phone 1140E are given below.

Location information

Effective Emergency 911 services also rely on accurate location information. The IP Phone 1140E supports the inclusion of the x-nt-location header in SIP messages that provide location information to a Nortel call server. The location information is selected by the user during registration, and must be correctly provisioned at the call server level (see appropriate NTP specific to your call server).

Dialing plan configuration

To allow operator control of disconnect during a 911 call, the IP Phone 1140E must identify a 911 call as soon as a 911 call is initiated. The IP Phone 1140E uses an emergency flag in the dialing plan to identify a 911 call. When the dialing plan detects that an emergency number is dialed, it automatically switches to operator controlled disconnect mode when the call is answered. The dialing plan can have multiple emergency numbers.

Examples of an emergency flag in the dialing plan

```
(911)|(911)# &&sip:$@$n;user=phone&&emergency
```

```
(4357)|(4357)# &sip:$@$n;user=phone&&emergency
```

Characteristics of E911 calls

During an E911 call, the user:

- cannot make outgoing calls
- is not notified of incoming calls and cannot accept incoming calls. Incoming calls receive a busy signal
- cannot transfer, join, or conference the E911 call, place the E911 call on hold, or park the E911 call
- cannot receive an incoming call
- cannot auto-retrieve a parked call and auto-retrieval of parked calls is not displayed
- cannot disconnect the E911 call. Only an E911 center or operator can disconnect the emergency call
- cannot change Audio Quality
- can reply to IM pop-ups, which are operational during an E911 call

During an E911 call the keys function as follows:

- the **Services**, **Inbox**, **Outbox**, and **Address-Book** keys are all disabled
- a right click of the mouse does not show the services menu
- the conspicuous keys are disabled
- the mute key and hold key are disabled
- the increase and decrease volume keys remain functional.
- the feature keys are visible and all except the speed-dial keys are functional.

IP Phone 1140E restrictions

Service package restrictions

Individual features and feature restrictions are sent to the IP Phone 1140E as a part of the service package every time a particular user logs on to the IP Phone 1140E. If the call server does not support service packages, or if the call server restricts some of the features in the service package, functionality of some features is restricted.

If functionality is restricted, the associated buttons and Context-sensitive soft keys are not accessible or do not respond.

Distinctive Ringing feature

The IP Phone 1140E does not support the CS 2000 and CS 2100 Distinctive Ringing feature.

NAT firewall traversal

The objective of putting devices behind a Network Address Translator (NAT) is to protect the devices from external interruption and to extend the public IP address space. However, the shield to stop unsolicited incoming traffic also has the drawback of breaking a number of IP applications, including SIP.

If a device is behind a NAT, transport addresses obtained are not publicly routable, and therefore, not useful in a number of multimedia applications. The limited lifetime of the NAT port mapping can also cause the SIP signaling to fail. If a port mapping is idle, it can be released by the NAT and reassigned to other applications.

The STUN protocol lets an IP Phone 1140E discover the presence and type of NATs between the IP Phone 1140E and the public Internet. In addition, an IP Phone 1140E can discover the mapping between the private IP address and port number and the public IP address and port number. Typically, a service provider operates a STUN server in the public Internet, with STUN-enabled IP Phones embedded in end-devices, which are possibly behind a NAT.

A STUN server can be located using DNS SRV records using the domain of the service provider as the lookup. STUN typically uses the well-known port number 3478. STUN is a binary encoded protocol with a 20-octet header field and possibly additional attributes. The STUN protocol learns the public IP addresses, and therefore, some security is necessary.

To initiate a STUN lookup, the IP Phone 1140E sends one or more Binding Request packets using UDP to the STUN server. These packets must be sent from the same IP address that the IP Phone 1140E uses for the other protocol, because this is the address translation information that the IP Phone 1140E tries to discover.

The server returns Binding Response packets, which tell the IP Phone 1140E the public IP address and port number from which it received the Binding Request. The IP Phone 1140E knows the private IP address and port number it used to send the Binding Request, and therefore, it learns the mapping between the private and public address space being performed by the NAT. If the Binding Response packets indicate the same address and port number as the request, the IP Phone 1140E knows no NATs are present.

The IP Phone 1140E supports two methods for NAT traversal of the signaling path:

- SIP_PING
- STUN

The NAT traversal method can be selected manually through the Device Settings menu or configured through the device configuration file. The default NAT traversal method is NONE.

The IP Phone 1140E can conduct SIP dialogs through a Symmetric NAT using UDP. This allows the phone to work from behind and/or in front of a symmetrical NAT with servers and/or clients that support RFC 3581. For this feature to work properly, the receiving end device must support RFC 3581. This feature is enabled or disabled through the USE_RPORT parameter in the device configuration file.

Note: RFC 3581 does not address NAT traversal for media or voice.

Three-port switch and VLAN functionality

System overview

The Full VLAN support feature can create telephone Voice-VLAN and PC Data-VLAN on the three-port switch of the IP Phone 1140E manually and automatically (see [Figure 12 "Voice-VLAN and Data VLAN" \(page 132\)](#)).

If both Data and Voice VLAN's are enabled on a three-port switch, only the frames with Data and Voice VLAN tagged go to networks. The IP Phone 1140E receives only the frames with Voice VLAN tagged and sends the frames with Voice VLAN tagged, while PC or Local Networks receive all kinds of frames.

When only voice VLAN is enabled on three-port switch, all kinds of frames go to the Network, the IP Phone 1140E receives only the frames with Voice VLAN tagged and send all frames with Voice VLAN tagged. PC or Local Networks receive all kinds of frames.

Figure 12
Voice-VLAN and Data VLAN

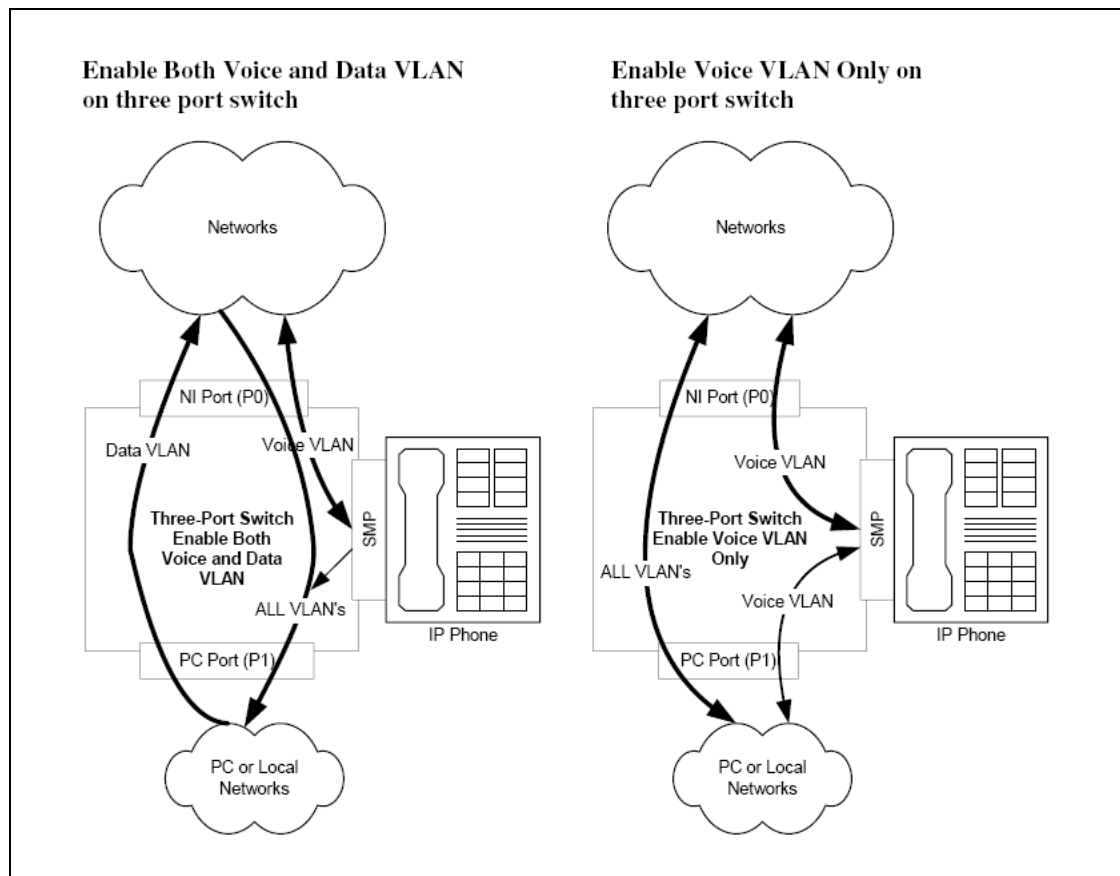


Table 57
Port functions on the three-port switch when VLAN is enabled

Ports	Voice VLAN enabled	Data VLAN enabled	Both Voice and Data VLAN enabled
Network Port (Port 0)	N/A	N/A	N/A
IP Phone 1140E Port (SMP)	Receiving the frames with Voice VLAN tagged only. Sending the frames with Voice VLAN tagged.	N/A	Receiving the frames with Voice VLAN tagged only. Sending the frames with Voice VLAN tagged.
PC Port (Port 1)	N/A	Tagging the incoming frame untagged and forwarding it to network port.	Tagging the incoming frame untagged and forwarding it to network port.

		Replacing the incoming frame tagged with VLAN other than Data-VLAN and forwarding it to network port. Sending all kinds of frames.	Replacing the incoming frame tagged with VLAN other than Data-VLAN and forwarding it to network port. Sending all kinds of frames.
--	--	---	---

VLAN configuration can be done either manually or through DHCP. Refer to ["Device Settings on the IP Phone 1140E with SIP Firmware" \(page 95\)](#) for more detail on configuring VLANs.

SIP messages supported by the IP Phone 1140E

SIP methods

Method	Supported?	Comments
INVITE	Yes	Mid-call re-invites for media changes also supported.
ACK	Yes	
BYE	Yes	
CANCEL	Yes	
OPTIONS	Response only	
INFO	Yes	Optionally used for in-session DTMF signaling, and Nortel call server specific NAT detection
PING	Yes	Proxy detection, monitoring and Nortel call server specific firewall traversal
REGISTER	Yes	For user registration
REFER	Yes	For transfer
NOTIFY	Yes	
SUBSCRIBE	Yes	
PUBLISH	Yes	For VQMon Publish

PRACK	Yes	No support for PRACK-specific early-media negotiation scenarios
MESSAGE	Yes	
UPDATE	Yes	UPDATE messages received in an early dialog state require reliable provisional responses. If PRACK is disabled, or not used by a local or remote party, some UPDATE operations will fail as described in RFC 3311. The support of UPDATE messages is not a configurable feature.

SIP responses

The following SIP responses are also supported:

- 1xx Response--Information Responses
- 2xx Responses--Successful Responses
- 3xx Response--Request Failure Responses
- 4xx Response--Server Failure Responses
- 6xx Response--Global Responses

1xx Response--Information Responses

1xx Response	Send	Receive	Comments
100 Trying	Yes	Yes	The IP Phone 1140E can generate this response for an incoming INVITE if it has taken too long to generate a 180 response. Upon receiving this response, the IP Phone 1140E waits for a 180 Ringing, 183 Session Progress, or 200 OK responses.
180 Ringing	Yes	Yes	The IP Phone 1140E begins local ringing through the active transducer.
181 Call is being forwarded	No	Yes	See 183.
182 Queued	No	Yes	See 183.
183 Session progress	No	Yes	The IP Phone 1140E accepts a 183 response with SDP to allow for early-media negotiation.

2xx Response--Successful responses

2xx Response	Send	Receive	Comments
200 OK	Yes	Yes	
202 Accepted	Yes	Yes	

3xx Response--Redirection responses

3xx Response	Send	Receive	Comments
300 Multiple Choices	No	Yes	When receiving this response, the IP Phone 1140E redirects the original request to next contact specified.
301 Moved permanently	No	Yes	When receiving this response, the IP Phone 1140E redirects the original request to the new contact specified. However, the IP Phone 1140E takes no additional special consideration of the "permanent" status of this change.
302 Moved temporarily	Yes	Yes	This response is sent to an incoming invite if the IP Phone 1140E has local call-forwarding enabled. When receiving this response, the IP Phone 1140E redirects the original request to the new contact specified.
305 Use Proxy	Yes	Yes	The IP Phone 1140E generates these responses when receiving requests that did not come through the configured SIP proxy. When receiving this request, the IP Phone 1140E contacts the new address in the Contact header field.
380 Alternate service	No	Yes	When receiving this request the IP Phone 1140E contacts the new address in the Contact header field.

4xx Response--Request failure responses

4xx Response	Send	Receive	Comments
400 Bad request	Yes	Yes	The IP Phone 1140E generates a 400 Bad Request response for various failure conditions generally when a request is invalid, and a more specific error response does not apply.
401 Unauthorized	No	Yes	Receiving a 401 response results in the IP Phone 1140E re-issuing the request using HTTP digest authentication.
402 Payment required	No	Yes	See default handling.
403 Forbidden	No	Yes	See default handling.
404 Not found	Yes	Yes	The IP Phone 1140E generates this response for requests to unknown users. Receiving this response falls through to the default handling.
405 Method not allowed	Yes	Yes	The IP Phone 1140E sends this response to a known method if it is received at a time when the IP Phone 1140E is not prepared to handle or the request is missing necessary information. Receiving this response falls through to the default handling.
406 Not acceptable	Yes	Yes	The IP Phone 1140E can send this response when receiving a REFER request which has an unsupported URI. Receiving this response falls through to the default handling.
407 Proxy authentication required	No	Yes	See 401.
408 Request timeout	No	Yes	See default handling.
410 Gone	No	Yes	See default handling.

413 Request entity too large	No	Yes	See default handling. The IP Phone 1140E does not automatically retry if a retry-after header is present.
414 Request---URL too long	No	Yes	See default handling.
415 Unsupported Media	Yes	Yes	The IP Phone 1140E can send this response when an incorrect content-type is detected for a request. Receiving this response falls through to the default handling.
420 Bad Extension	Yes	Yes	The IP Phone 1140E can respond with a 420 when checking required extensions of incoming requests. When receiving a 420, see default handling. The IP Phone 1140E does not retry the request.
480 Temporarily unavailable	No	Yes	See default handling.
481 Call leg/ transaction does not exist	Yes	Yes	Incoming requests are matched against existing dialogs. If a request appears to be in-dialog, but does not have an existing dialog, the IP Phone 1140E responds with a 481. For incoming 481 responses, the default handling is used.
482 Loop detected	Yes	Yes	Default handling is used when this response is received.
483 Too Many Hops	No	Yes	See default handling.
484 Address Incomplete	No	Yes	See default handling.
485 Ambiguous	No	Yes	See default handling. The IP Phone 1140E does not attempt to retry the request.

486 Busy Here	Yes	Yes	The IP Phone 1140E can respond with this if the user is on the IP Phone 1140E, and the IP Phone 1140E has reached its maximum number of allowed calls and cannot present the incoming call to the user. When this message is received by the IP Phone 1140E an error is displayed and a busy tone is played.
487 Request Canceled	Yes	Yes	See default handling.
488 Not Acceptable	Yes	Yes	The response is used by the IP Phone 1140E when a failed media negotiation occurs.
491 Request Pending	Yes	Yes	The IP Phone 1140E sends and receive this message in GLARE conditions.

5xx Response--Server failure responses

5xx Response	Send	Receive	Comments
500 Internal Server Error	Yes	Yes	The IP Phone 1140E can send this response when a request is received but the IP Phone 1140E software is not in a correct state to handle it. When receiving this message the IP Phone 1140E displays an error for the user.
501 Not Implemented	No	Yes	See default handling.
502 Bad Gateway	No	Yes	See default handling.
503 Service Unavailable	Yes	Yes	
504 Gateway timeout	No	Yes	See default handling.
505 Version Not Supported	Yes	Yes	

6xx Response--Global responses

6xx Response	Send	Receive	Comments
600 Busy Everywhere	Yes	Yes	The IP Phone 1140E can send this response when the IGNORE setting is set to NETWORK, and the user chooses to ignore an incoming call. When received, this response falls through the default handling.
603 Decline	Yes	Yes	The IP Phone 1140E can send this response when the user declines an incoming call. An optional reason can be supplied.
604 Does Not Exist Anywhere	No	Yes	See default handling.
606 Not Acceptable	No	Yes	See default handling.

Default error handling

All 4xx/5xx/6xx responses (with the exception of 401/407) received by the IP Phone 1140E when attempting to initiate a call result in the display of an error on the screen, and typically results in fast or regular busy tone.

If a media negotiation fails during dialog setup, the IP Phone 1140E terminates the dialog.

If an in-dialog failure occurs during media (re)negotiation, the IP Phone 1140E falls back to previously negotiated media settings. When a failure occurs that makes this impossible, the IP Phone 1140E attempts to clear the call by terminating the dialog.

SIP header fields

The following table contains the supported SIP headers.

Header field	Supported?
Accept	Yes
Accept-Encoding	Yes
Accept-Language	Yes

Alert-Info	Yes
Allow	Yes
Allow-Events	Yes
Authentication-Info	Yes
Authorization	Yes
Call-Id	Yes
Call-Info	Yes
Contact	Yes
Content-Disposition	Yes
Content-Encoding	Yes
Content-Length	Yes
Content-Type	Yes
Cseq	Yes
Date	Yes
Expires	Yes
Error-Info	Yes
Max-Forwards	Yes
Mime-Version	Yes
Organization	Yes
P-Access-Network-Info	Yes

P-Asserted-Identity	Yes
P-Associated-URI	Yes
P-Called-Party-ID	Yes
P-Charging-Function-Addresses	Yes
P-Charging-Vector	Yes
P-Media-Authorization	Yes
P-Preferred-Identity	Yes
P-Visited-Network-ID	Yes
Path	Yes
Priority	Yes
Privacy	Yes
Proxy-Authenticate	Yes
Proxy-Require	Yes
RAck	Yes
Reason	Yes
Record-Route	Yes
Refer-To	Yes
Referred-By	Yes
Remote-Party-ID	Yes
Replaces	Yes

Reply-To	Yes
Require	Yes
Resource-Priority	Yes
Retry-After	Yes
Route	Yes
RSeq	Yes
Server	Yes
Service-Route	Yes
Subject	Yes
Supported	Yes
Timestamp	Yes
To	Yes
Unsupported	Yes
User-Agent	Yes
Via	Yes
Warning	Yes
WWW-Authenticate	Yes

Session description protocol usage

SDP Headers	Supported?
v--Protocol version	Yes

o--Owner or creator and session identifier	Yes
s--Session name	Yes
t--Time description	Yes
c--Connection information	Yes
m--Media name and transport address	Yes
a--Media attribute lines	Yes

SDP and Call Hold

The IP Phone 1140E can support sending and receiving of hold using the method specified by RFC2543 and RFC3261/3264.

Transport layer protocols

Protocol	Supported?
Unicast UDP	Yes
Multicast UDP	No
TCP	No

SIP security authentication

Authentication	Supported?	Comments
Digest Authentication	Yes	
Proxy-to-User Authentication	Yes	
User-to-User Authentication	No	The IP Phone 1140E responds to a 401, but never challenges incoming requests with a 401 response.

S/MIME	No	
AKA	No	

SIP DTMF Digit transport

Transport type	Supported?
RFC 2833	Yes
In-band tones	Yes
Out-of-band tones	Yes (vnd.nortelnetworks.digits)

Supported subscriptions

Subscription type	Supported	Nortel Call Server specific
address-book	Yes	Yes
call-park	Yes	Yes
dialog	Yes	Yes
presence	Yes	Yes
message-summary	Yes	No
ua-profile	Partial	Yes
service-package	Yes	Yes
network-redirection-reminder	Yes	Yes

Supported instant messaging

Message type	Supported?
plain text	Yes

Nortel unencrypted	Yes
Nortel encrypted	Yes

Maintenance and troubleshooting

Convert SIP Firmware to UNISTim Firmware

The IP Phone 1140E can be ordered with UNISTim Firmware installed or with SIP Firmware installed. If you have an IP Phone 1140E with UNISTim Firmware, and you convert the firmware from UNISTim to SIP, the UNISTim Firmware is overwritten. To convert an IP Phone 1140E from SIP Firmware to UNISTim Firmware, a firmware reload is required.

Procedure 14

Reloading UNISTim Firmware

Step	Action
1	<p>Determine the appropriate UNISTim version to match the hardware release number of your IP Phone 1140E.</p> <p>There are different versions of UNISTim Firmware available for download. Which version you choose depends on the hardware release number of your particular IP Phone 1140E.</p> <p>If the hardware release number of your IP Phone 1140E is among the following hardware release numbers, download UNISTim Firmware version release 0625C39 or higher (the hardware release number is the Product Engineering Code [PEC] followed by the release number):</p> <ul style="list-style-type: none"> • NTYS05ACE6 20 • NTYS05BCE6 20 • NTYS05BCGSE6 04 <p>If the hardware release number of your IP Phone 1140E is not among the previous list, download UNISTim version release 0625C23 or higher.</p> <p>Note: UNISTim Firmware version names contain numbers and letters. Use the last three characters in a version to determine the minimum required version for the conversion. For example, C39 is greater than C23.</p>

- 2 Download the appropriate UNISTim Firmware file to your TFTP server.
- 3 Create an 1140eSIP.cfg file containing the following information:
[FW]
DOWNLOAD_MODE FORCED
VERSION xxx
FILENAME yyy.bin

where xxx is the UNISTim version number appropriate for the hardware release of your IP Phone 1140E, for example, 0625C23, and yyy.bin is the filename containing the version number, for example, 0625C23.bin.
- 4 Power the IP Phone 1140E off and on. The IP Phone 1140E reboots and contacts the TFTP server upon bootup and downloads the new UNISTim Firmware.

—End—

IP Phone 1140E diagnostics

Network-related issues can be debugged using Network Diagnostic Utility (NDU) built into the IP Phone 1140E.

Another way to diagnose a problem on an IP Phone 1140E is to capture a message trace using any appropriate software.

The IP Phone 1140E has Problem Determination Tools (PDT). These can be accessed through a Telnet session using the IP address of the IP Phone 1140E (for login and password, contact Nortel).

Problem: Server unreachable after the IP Phone 1140E is powered up.

If the display indicates that the server is unreachable and it continuously resets, some parameters must be configured. Things to consider when setting parameters:

- Enter requested information in the menu fields by pressing the number keys on the dialpad. Press the asterisk (*) key to enter a period (.) when entering an IP address.
- To record the entry and advance the initialization to the next parameter, press **OK**.
- To abandon the manual configuration process and restart the power-up, press **Cancel**.

- To manually enter parameters, use the **BKSpace** or **Clear** Context-sensitive soft keys to edit the default entry. BKSpace deletes each character as the key is pressed. Clear deletes the entire entry.
- Each parameter must have a corresponding entry. An audible beep indicates a field entry must be made before advancement to the next parameter.

Problem: Firmware Download Failure

If you are having trouble downloading firmware, review the following bullets.

- Is the TFTP IP address correct within the 1140E device settings menu?
- Is the IP Phone 1140E connecting to the TFTP server log?

Check any firewall configuration settings to allow TFTP protocol access.

Is the syntax within the 1140e.cfg or 1140eSIP.cfg correct? See Provisioning file supported sections in the chapter "[Configure the provisioning server](#)" (page 29). Supported sections describe the syntax of the configuration file.

- Does DOWNLOAD_MODE = AUTO and is VERSION less than the current running firmware version? If a file does not download using the AUTO selection, it is possible the version number is not high enough. A version number exists permanently on the IP Phone 1140E until a higher version number is downloaded through the device configuration file or you select **Srvcs, System, Erase User Data** on the IP Phone 1140E.
- Check FILENAME. Does this exist on the TFTP server?
- Check to make sure your firewall settings allow for the provisioning protocol (TFTP, FTP, OR HTTP) to go through.

There is a chance of firmware download failure, leaving the IP Phone 1140E with no valid application code and only the boot loaders. If this happens, the boot loaders are executed and handle the application download.

Problem: Firmware Conversion Failures

There are four different boot loaders in the FLASH and application load. Various boot loaders are used to recover the IP Phone 1140E in case of any failure occurs.

- If a conversion fails before anything is written to FLASH, the IP Phone 1140E reboots with the UNISlim Firmware load.
- If the firmware download fails while the application is being written to FLASH, there are two possible recovery methods:
 - If an application file was not created, after power up the IP Phone 1140E jumps to the BootC loader and downloads a new application load using the same mechanism as the application.

- If the application is executed and the file created is corrupted, the IP Phone 1140E crashes. In this case, force the IP Phone 1140E to use BootC by holding up key and “2” during power up.

Problem: Users of the IP Phone 1140E complain that their banner is not updated with their custom banner

When the banner is set to FORCED in the device configuration file, the user's banner is overwritten by the value in the device configuration file.

Problem: Provisioning Error is displayed on the IP Phone 1140E display.

The Provisioning Error is displayed on the screen when the IP Phone 1140E is unable to contact the TFTP, FTP, or HTTP server.

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Publication: NN43113-300
Document status: Standard
Document version: 03.09
Document date: 15 September 2008

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